MOS-based Handover Protocol for Next Generation Wireless Networks

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***Abstract*—Next Generation Wireless Networks (NGWNs) are expected to provide high data rate and optimized quality of service to multimedia and real-time applications over the Internet Protocol (IP) networks. To achieve these goals, handover plays a very critical role in maintaining the seamless connectivity when mobile terminals move across different cells or networks. In this paper, we propose a novel scheme compliant with the IEEE 802.21 standard for handover in an integrated scenario with UMTS and WiMAX networks. We use the call quality, measured using Mean Opinion Score (MOS), as the major metric for handover optimization. We compare the proposed MOS-based handover scheme with the traditional RSS-based handover scheme. The numerical results demonstrate that our proposed scheme can maintain high call quality and reduce the probabilities for both handover dropping and call dropping.**

***Index Terms*—VoIP, WiMAX, UMTS, QoS, Next Generation Wireless Network.**

I. Introduction

The envisaged NGWNs will integrate a number of different networks, such as Universal Mobile Telecommunications System (UMTS) and Worldwide Interoperability for Microwave Access (WiMAX) to provide a comprehensive and secure all-IP based services to mobile terminals. Future mobile terminals will be equipped with multiple network interface cards, which enable the mobile users to connect to different networks and access any service anywhere and anytime. However, heterogeneous networks are different in data rate, traffic classes, call admission mechanisms, etc. How to seamlessly transfer user service between networks of the same type or between different networks is a well-known handover issue, and has become one of the major challenges in developing and deploying the NGWNs.

With the rapid growth of wireless packet-switched networks, sending data through the Internet rather than the Public Switched Telephone Network (PSTN) has become a better option in terms of cost for both users and service providers. This has led to enormous growth of real-time applications based on Voice over IP (VoIP), enabling the mobile users to make calls through internet anywhere and anytime with better communication quality and less cost than PSTN. With a growing number of moving users, it has become a necessity to guarantee the Quality of Service (QoS) for applications that demand more bandwidth, better network connectivity and seamless handover. Moreover, wireless networks are susceptible to delay, packet loss and poor call quality due to the low Signal to Interference and Noise Ratio (SINR). An efficient handover management scheme should be designed to achieve better call quality in NGWNs.

In the existing handover schemes, a handover is generally triggered by either the detection of degradation in Received Signal Strength (RSS) or using other metrics such as measurement from network load, power consumption, user preference and available bandwidth. The traditional handover protocols based on RSS or cost functions [1] have flaws and are not competitive enough to achieve satisfactory QoS. The mobile terminal has to scan continuously for the current and available networks signal strength. This scanning procedure utilizes wireless resources and also encounters with the wireless channel access delay. Mobile terminals with continuous scanning also consume more battery power, thereby resulting in energy inefficiency. Another problem is the fading signal which will give rise to the ping-pong effect, resulting in unnecessary handover [2].

In this paper, we investigate the following problem: *for an urban area deployed with both UMTS and WiMAX base stations, when a mobile terminal experiences degrading call quality from its current connection, how can we choose an optimal attachment point for the mobile terminal to handover in terms of maximizing the call quality measured using Mean Opinion Score (MOS)?* We model this problem as an optimization problem by considering the available bandwidth at the base stations, the communication delay and loss, and the MOS values. A centralized algorithm is designed to compute the optimal base station for handover. To enable the handover between base stations both in the same network and in different networks, we design a handover protocol compliant with the recently proposed IEEE 802.21 standard, which is also called as the Media Independent Handover (MIH) [3]. The standard defines a media-independent handover framework that can significantly reduce the complexity for handover between heterogeneous network technologies. We have done extensive numerical simulations to evaluate our MOS-based handover protocol. Simulation results show that our scheme can provide much better performance than the traditional RSS-based handover schemes.

The rest of the paper is organized as follows. Section II briefly discusses the related work. Section III presents the problem formulation. Section IV gives the optimal base station selection algorithm. Section V describes the handover protocol design. Section VI discusses the numerical results. Finally, in Section VII we conclude this paper.

II. Related work

Most of the existing work on handover in UMTS, WLAN and WiMAX is based on bandwidth [4], SINR [5] or RSS [6] [7]. Yang et al in [8] proposed that, when roaming from WiMAX networks to Wi-Fi networks, it is reasonable to initialize handover to Wi-Fi when Wi-Fi is available because Wi-Fi networks can provide high bandwidth and lower cost. However, they do not consider the handover probability. When the user is moving, i.e. when the handover is required, Wi-Fi network can be very small and a user might need to handover again, thus increasing the handover probability and affecting QoS. In [5], a handover algorithm is proposed to use the received SINR from various access networks as the handover criteria. There are different environmental and networking factors which causes variation in SINR. This results in increase in the handover probability and might cause unnecessary handover. Paper [9] is based on forced termination of calls due to handover failure. The dropping of a handover call is generally considered more serious than blocking of a new call. Therefore, a certain amount of bandwidth (also called guard channels) is exclusively reserved for handovers. This amount of bandwidth can be either fixed or adaptively controlled with respect to the current traffic load. RSS and bandwidth are important factors but there are several other factors which might degrade the quality of voice signal. A user might be just standing beside the base station and there might be sufficient bandwidth available, but the network to which the user is attached might not support VoIP call well, or there might be other network available which might provide better quality. In our scheme, we not only consider bandwidth but also take into consideration the quality of service parameters.

III. Problem formulation

Consider an urban area where a UMTS network and a WiMAX network coexist, as shown in Figure 1. We use the tightly coupled architecture [10], in which a single Radio Network Controller (RNC) maintains the network information such as the available capacity at each base station and the quality of each wireless connection. This can be achieved by requesting each base station to periodically update the resource usage and the quality of currently served applications. All handovers occurred in this area are managed and optimized at the RNC.

As shown in Figure 1, the connection setup for communication between the mobile terminals $MT1$ and $MT2$ can be divided into two parts: connection between the base stations, and connection between the mobile terminal and base stations. The connection between the base stations goes through the Internet using the wired medium, and voice data is transmitted using the VoIP protocol. Since the Internet commonly has large communication bandwidth, it can provide relatively stable communication quality, thus having little impact on the voice quality. However, the communication between mobile terminal

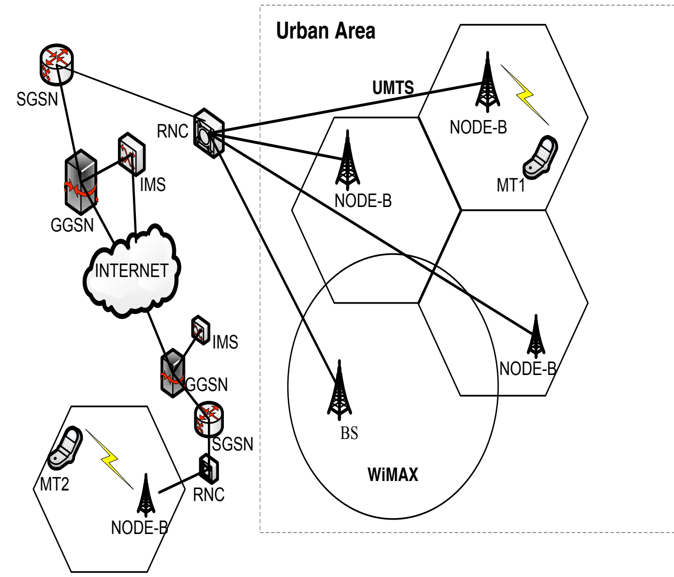

Fig. 1. Tightly coupled architecture for an integrated scenario

and base stations is wireless. Since wireless channels are prone to errors due to noise interference as well as the movement of the mobile terminal, the quality of the wireless channel generally dominates the quality of the VoIP call. In this study, we focus on the communication between the mobile terminals and the base stations.

Let $B = \{b_1, b_2, \ldots, b_n\}$ be the set of UMTS and WiMAX base stations deployed in urban area, and $M = \{m_1, m_2, \ldots, m_n\}$ be the set of mobile terminal. Given a base station b_i and a mobile terminal m_j, let $p_{i,j}$ and $d_{i,j}$ denote the packet loss rate and the average packet delivery delay between b_i and m_j, respectively. If mobile terminal m_j is not in the coverage range of base station b_i, $p_{i,j} = 1$ and $d_{i,j} = \infty$. Consider a mobile terminal m_i which is experiencing poor call quality, our objective is to design an efficient solution to select the best base station for m_i to handover, by which the call quality is maximized.

Mean Opinion Score (MOS) is one of the major metrics for evaluating the quality of a VoIP call, and will be the main metric used in our work for handover optimization. Let $M_{i,j}$ denote the MOS value for the VoIP call between base station b_i and mobile terminal m_j. In [11], MOS is computed as follows:

$$M_{i,j} = 1 + 0.035R + 7 * 10^{-6} R(R-60)(100-R) \quad (1)$$

where $R = 94.2 - I_e - I_d$. I_e is the impairments caused by different types of losses occurred due to codecs and network. In the E-model proposed in [12], I_e is modeled as follows:

$$I_e = \gamma 1 + \gamma 2 * ln(1 + \gamma 3 * p_{i,j}) \quad (2)$$

where $p_{i,j}$ denotes the packet loss rate between base station b_i and mobile terminal m_j. For a given Codec, $\gamma 1$, $\gamma 2$, $\gamma 3$ are constants, e.g., for G.711 $\gamma 1$ is 0, $\gamma 2$ is 30 and $\gamma 3$ is 15 [12].

I_d represents the impairment caused by delay particularly mouth-to-ear delay. The I_d for a VoIP steam is given by

$$I_d = 0.024 * d_{i,j} + 0.11(d_{i,j} - 177.3)H(d_{i,j} - 177.3) \quad (3)$$

where $H(x) = 0$ if $x < 0$; otherwise $H(x) = 1$. $d_{i,j}$ denote the average packet delivery delay between base station b_i and mobile terminal m_j. It is the delay which is composed of three components: codec delay, payout delay, and network delay.

In VoIP applications, the call quality is traditionally measured from a user's perception using MOS in a range varying from 1 (bad) to 5 (excellent) [13] [14]. The relation between the R-factor and the MOS rating is given by [15],

$$MOS = \begin{cases} 1, & \text{For} \quad R < 6.5, \\ M_{i,j} \text{ given by Equation (1)}, & \text{For} \quad 6.5 \leqslant \text{R} \leqslant 100, \\ 4.5, & \text{For} \quad R > 100 \end{cases} \quad (4)$$

By Equations (1), (3) and (2), it can be seen that $M_{i,j}$ can be expressed as a function of packet loss $p_{i,j}$ and delay $d_{i,j}$, i.e., $M_{i,j} = f(p_{i,j}, d_{i,j})$. Suppose that mobile terminal m_j is currently making a VoIP call demanding bandwidth of $\bar{c}$ and experiences poor call quality, our goal is to choose the best base station that meets the bandwidth requirement for m_j to handover in terms of maximizing $M_{i,j}$. The base station selection problem can be formulated as the following optimization problem:

$$\begin{array}{ll} \text{maximize} & M_{i,j} = \max_{b_i \in B} f(p_{i,j}, d_{i,j}) \\ \text{s.t.} & c_i \geqslant \bar{c}; \end{array} \quad (5)$$

where c_i represents the available bandwidth capacity at the base station b_i.

IV. Optimal Base Station Selection

In this section, we present the solution for choosing the optimal base station that maximizes the MOS value, assuming that the RNC has the knowledge of the bandwidth capacity of each base station, and the delay and packet loss rate for each wireless link between mobile terminal and base stations. The details on how to obtain these parameters will be described in next section.

From Equation (4), it can be seen that MOS monotonously increases with the increase of R when $0 < R < 100$. By Equations (1), (2) and (3), it is easy to prove that the MOS value monotonously increases with the decrease of packet loss rate and packet delay. Then we have the following observation.

Observation: Given a mobile terminal m_j and two base stations $b_i : (p_{i,j}, d_{i,j})$ and $b_k : (p_{k,j}, d_{k,j})$. If $p_{i,j} \leqslant p_{k,j}$ and $d_{i,j} \leqslant d_{k,j}$, we have $M_{i,j} \geqslant M_{k,j}$.

The above observation enables to quickly drop unsuitable candidates during base station selection process. Let $b_k : (p_{k,j}, d_{k,j})$ be the current severing base station for m_j. The base station selection procedure works as follows: we initially use $p_{k,j}$ and $d_{k,j}$ as the benchmark for the base station selection. Given a base station $b_i : (p_{i,j}, d_{i,j})$ in B,

1) If $p_{i,j} \geqslant p_{k,j}$ & $d_{i,j} \geqslant d_{k,j}$, b_i can not provide better call quality than the current serving base station b_k according to the observation.
2) If $p_{i,j} \leqslant p_{k,j}$ & $d_{i,j} \leqslant d_{k,j}$, b_i can provide better call quality than b_k. We use $b_i : (p_{i,j}, d_{i,j})$ as a new benchmark to continue base station selection.
3) If $p_{i,j} \geqslant p_{k,j}$ & $d_{i,j} \leqslant d_{k,j}$ or $p_{i,j} \leqslant p_{k,j}$ &$d_{i,j} \geqslant d_{k,j}$, it is hard to judge directly which one is better. The MOS values will be computed and used for base station selection.

Let B_j be the set of base stations which satisfy the bandwidth requirement for the mobile terminal m_j. The detailed algorithm for base station selection is given in Algorithm 1.

input : $B_j = \{b_i : (p_{i,j}, d_{i,j})\}$, b_k, m_j
output: optimal b^*

$p = p_{k,j}$; $d = d_{k,j}$; $M = M_{k,j}$; $b^* = b_k$;
for *each* $b_i : (p_{i,j}, d_{i,j})$ *in* B_j **do**
  if $(p_{i,j} \geqslant p)\&(d_{i,j} \geqslant d)$ **then**
    continue;
  else if $(p_{i,j} \leqslant p)$***and***$(d_{i,j} \leqslant d)$ **then**
    $p = p_{i,j}$; $d = d_{i,j}$; $M = M_{i,j}$, $b^* = b_i$;
  else if $(p_{i,j} \geqslant p\&d_{i,j} \leqslant d)$***or***$(p_{i,j} \leqslant p\&d_{i,j} \geqslant d)$ **then**
    Calculate $M_{i,j}$;
    if $(M_{i,j} \geqslant M)$ **then**
      $p = p_{i,j}$; $d = d_{i,j}$; $M = M_{i,j}$, $b^* = b_i$;
    end
  end
end

Algorithm 1: Optimal base station selection

Let $|B_j|$ be the number of base stations in B_j. The time complexity of Algorithm 1 is $|B_j|$. The proposed MOS-based base station selection scheme has several advantages in comparison with the existing solutions. Firstly, the algorithm guarantees that the selected base station must meet the requirement on bandwidth, thus avoiding frequent handover failures as in simple RSS-based solutions. Secondly, the proposed scheme is energy efficient since there is no need for continuous scanning. In RSS-based solutions, the mobile node has to continuously scan the current available networks, which consumes quite a lot battery power. Whilst in our scheme, the RNC will be responsible for collecting QoS parameters and making handover decisions. Thirdly, our scheme can avoid unnecessary handover. In situations where a user is having low RSS but quality of voice call from the user point of view is good, the RSS-based handover protocol takes decision to do handover even if there is no need to do handover as quality of the call is acceptable.

V. Handover protocol design

This section presents our MOS-based handover protocol designed based on the IEEE 802.21 framework [3]. We will first give a brief overview of the IEEE 802.21 standard, and then describe the details of our handover protocol.

A. IEEE 802.21 Standard

The IEEE 802.21 standard defines a media-independent handover (MIH) framework that can significantly improve seamless handover between heterogenous network technologies. IEEE 802.21 facilitates the handover between different radio access technologies without call interruption, providing seamless connectivity for the mobile terminal, and improving the quality of service. MIH framework is based on a protocol stack implemented in all the devices involved in the handover, and provides a common interface for the link layer functions which is independent of radio access technologies. It consists of a MIH client which sits at user equipment end. MIH server resides in the core network. Handover decision for all the users in that zone is based on the information provided by MIH. In the IEEE 802.21 standard, Media Independent Handover Functions (MIHFs) are defined to provide a generic link layer.

The MIH framework provides a group of MIH functionalities that facilitate both mobile-initiated and network-initiated handovers. MIH provides a framework which exchanges the events, commands and information about QoS parameters, current link layer conditions and traffic load with different radio access technologies, which are used as input for taking decision for handover. The major components include:

- MIH function (MIHF), which is a logical entity that provides abstract services to the higher layers through a media independent interface and obtains information from the lower layers through media specific interfaces. It provides three types of services: (1) Media-Independent Event Service (MIES) for detecting and reporting changes in link layer properties; (2) Media-Independent Command Services (MICS) for local or remote MIH users to control link state; and (3) Media-Independent Information Service (MIIS) for providing information about neighboring networks.
- Service Access Points (SAPs), which defines both media-independent and media-specific interfaces. It includes: (1) MIH_SAP for high layers to control and monitor different links; (2) MIH_LINK_SAP for MIHF to control media-specific links; (3) MIH_NET_SAP to support the exchange of MIH information and messages with the remote MIHF.

For the details of the framework, refer to [3][16].

B. Parameter Acquisition

To perform the MOS-based handover, our protocol needs the following information: the set of candidate base stations, the available bandwidth capacity at each candidate base station, the delay and packet loss rate for each wireless link between the mobile terminal to a candidate base station, and the MOS of the current connection. All these information can be obtained using the MIH functions provided in the IEEE 802.21 framework, as described below.

Candidate base stations: The neighboring base stations information can be collected using the Media-Independent Information Service (MIIS) in IEEE 802.21. The intelligent MIH connection monitoring manager sits between the application and the device radio modem to monitor the wireless access, network status and availability. Link manager is responsible for managing local link. It controls the local link by responding to MIH commands.

Bandwidth: Each base station keeps track of its available bandwidth capacity. As all base stations are wired to the RNC, the available bandwidth capacity of each base station can be reported to the RNC at a regular time internal, or can be retrieved by the RNC dynamically.

Delay and packet loss: When a VoIP call is made from any device (mobile, laptop, iphone etc), it travels through the mobile terminal, NodeB (base station), RNC, SGSN, Gateway GPRS Support Node (GGSN) and Internet. The only air interface is between the mobile terminal and the base station, which commonly has a significant effect on the call quality. In a heterogeneous network, each mobile terminal is equipped with multiple radio receivers. The quality of the links from the mobile terminal to different networks can be monitored using the MIES and the SAPs functions. Link manager detects the link quality of the current call. Connection monitoring manager detect the link quality of the available base station. To calculate the value of delay, packet loss and MOS of the candidate base station, there are two different approaches: 1) Calculate the packet loss, delay and MOS by sending test packets from mobile terminal to other available base stations. 2) Establish multiple tunneling between mobile terminal and base stations at a time [17]. With multiple tunneling approach, a mobile terminal has to establish tunneling with available base stations. We use the first approach as multiple tunneling causes an extra overhead on the mobile terminal.

MOS: The MOS value of the current ongoing call is used to initiate the handover process and to compare with other potential connections. This information is directly available as there is an ongoing connection between the mobile terminal and the serving base station. For the MOS values of other potential connections, they can be estimated based on the delay and packet loss rate using Equation (1).

C. MOS-based Handover Protocol

In our protocol, the handover is triggered by the mobile terminal, whereas the decision on whether the handover will be finally performed and how the handover is performed are made at the RNC. The mobile terminal monitors the call quality and the current link state. If the mobile terminal detects the MOS value of the current ongoing call is below a predefined threshold, it sends a request to the RNC for handover. Once the RNC receives the request message, it will send a query to the candidate base stations. If there is another base station that can provide better service to the mobile terminal, handover will be immediately executed; otherwise the handover request is rejected. Figure 2 shows the flowchart of the protocol we designed, which consists of three steps: handover request, base station selection, and handover execution.

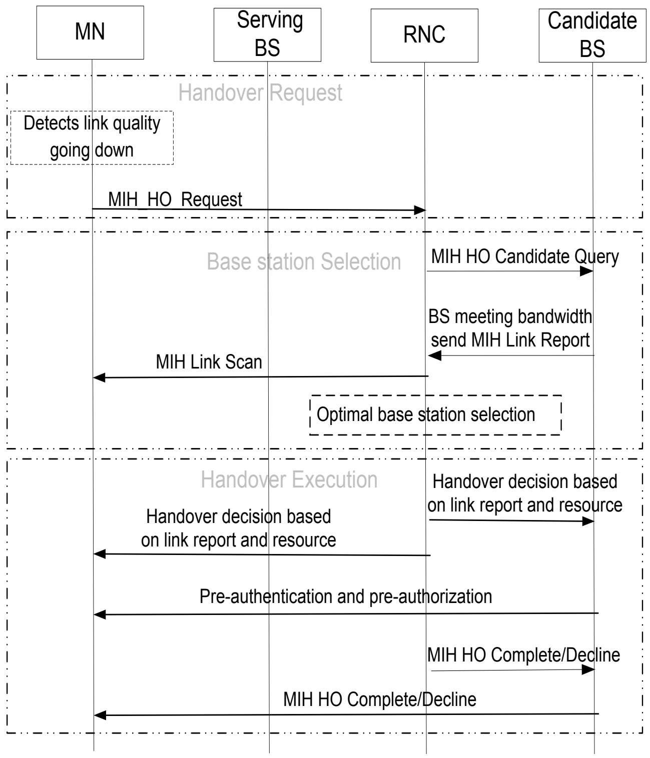

Fig. 2. Handover design Protocol

Handover Request: When a mobile terminal m_j served by the base station b_i detects that the MOS of the current ongoing VoIP call is below the threshold, it sends an MIH_HO_Request message to the RNC, and this message contains the following information: (1) the current MOS value $M_{i,j}$ (2) the current link status $\{p_{i,j}, d_{i,j}\}$; and (3) the bandwidth requirement c_j.
Base Station Selection: Once the RNC receives an MIH_HO_Request message from mobile terminal m_j, it broadcasts an MIH_HO_Candidate_Query message encapsulating the required bandwidth c_j to all the base stations. The RNC also sends an MIH_Link_Scan message to the mobile terminal m_j to initiate the process for measuring the quality of the wireless links from the mobile m_j to the other base stations. Only the base stations which have available bandwidth no smaller than the requested bandwidth c_j, will join the link measurement process. After link quality measurement, each base station b_k which has available bandwidth no smaller than c_j sends an MIH_Link_Report message to report $(p_{k,j}, d_{k,j})$ to the RNC, and b_k will reserve the bandwidth c_j for mobile terminal m_j. When the RNC collects all the link reports from the suitable base stations, Algorithm 1 is executed to compute the optimal base station for handover.
Handover Execution: If there is no other base station that can provide better call quality than the current serving base station, the RNC sends an MIH_HO_Decline message to the mobile terminal to terminate the handover process, and sends another message to the other base stations to release the bandwidth resource reserved for mobile terminal m_j; otherwise the detailed steps are performed for handover from the current serving base station to the new base station. When the handover is completed at the higher layers, a MIH_HO_Complete message to the MIH.

VI. Simulation and Results

To evaluate the proposed scheme, we implemented our MOS-based handover protocol in MATLAB. We simulated it in an integrated environment with WiMAX and UMTS networks, and compared its performance with the RSS-based handover [18]. In this study, we measure MOS, Handover Dropping Probability (HDP) and Call Dropping Probability (CDP).

A. Simulation Setup

In our simulation, we deploy 10 UMTS base stations and 4 WiMAX base stations in a $10000m*10000m$ area. The mobile terminals are uniformly placed in the UMTS or WiMAX cells. Each of the UMTS or WiMAX cell has a base station. All the base stations are connected to the RNC where the handover algorithm is located. The diameter of a UMTS cell is configured to 2 km, and the diameter of a WiMAX cell is configured to 3 km. When a mobile user makes a VoIP call, the voice packets are carried from mobile terminal to the RNC through Node-B. Even though there are different codecs such as G.711, G.721, G.722, etc. We use G.711 since it has the least compression delay [19]. Each simulation is run for 10 minutes.

In our simulation, we use a 2D random walk model to simulate the movement of the mobile terminals. Because some mobile terminals are believed to move in an unexpected way, random walk mobility model is proposed to mimic their movement behavior [20]. The random walk model is a stateless mobility process, where the information about the previous status is not used for the future decision. That is, the current parameter information is independent with its previous parameter information. The movement of each mobile terminal is controlled by two parameters: the moving direction θ and the step size L. Each time the mobile user walks for the distance L, a new direction is randomly chosen (i.e. four possibilities: 1) forward, 2) backward, c) left and d) right) then walks for another distance L.

We compare our MOS-based handover protocol with RSS-based handover protocol proposed in [21]. The RSS is calculated using the following function

$$RSS = -62.5 - 26.5 * log_{10}(d) \tag{6}$$

where d is the distance between a mobile terminal and a base station. If mobile terminal detects the RSS value below the threshold, the mobile terminal scans for the available networks and handovers the call to the base station providing higher RSS. If the mobile terminal fails to find a better base station, the handover request is rejected, and the mobile terminal continues the call by connecting to the same base station.

B. Simulation Result

1) Mean Opinion Score (MOS): In this set of simulations, we use only one mobile terminal and monitor the MOS during its movement. The MOS threshold for MOS-based handover scheme is set to 3.0. Figure 3 shows the MOS values for the proposed MOS-based handover scheme. The handovers occurred are marked by numbers on the graph. Initially, the mobile terminal has a MOS value of 3.6. Until the end of the first minute, the mobile terminal maintains the connection with the current serving base station. The mobile terminal detects the decline in quality after the first minute and performs a search operation looking for the base station with highest MOS among the available base stations for handover. Since the mobile terminal did not have a base station with a stable and better MOS value, it continues the service with the current base station until it finds a suitable base station approximately until two minutes. Once the mobile terminal is able to get a base station with stable MOS value higher than the threshold, it performs a handover which is numbered two. From the graph, we can see that MOS value of the call is maintained and the mobile terminal tends to select the base station which provides better MOS value every time it decides to make a handover due to degradation of the call quality, and thus guarantees the call quality.

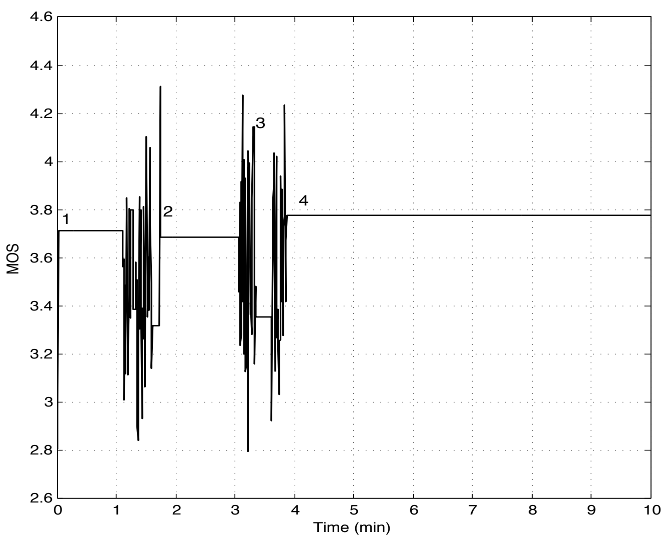

Fig. 3. MOS-Based Scheme

Figure 4 displays the MOS value for the RSS-based handover scheme, where the RSS threshold for handover is configured to -68dBm. The mobile terminal starts with an initial value for MOS of 3.6 and continues the connection with the current base station until it realizes a drop in the RSS value below the threshold. The mobile terminal performs a scan for the target base stations with better RSS. Since the RSS-based handover scheme does not consider the MOS value of the target base station, the probability of choosing a base station with higher RSS but less MOS value is higher. For example, it can be seen from the figure that the first handover of the mobile

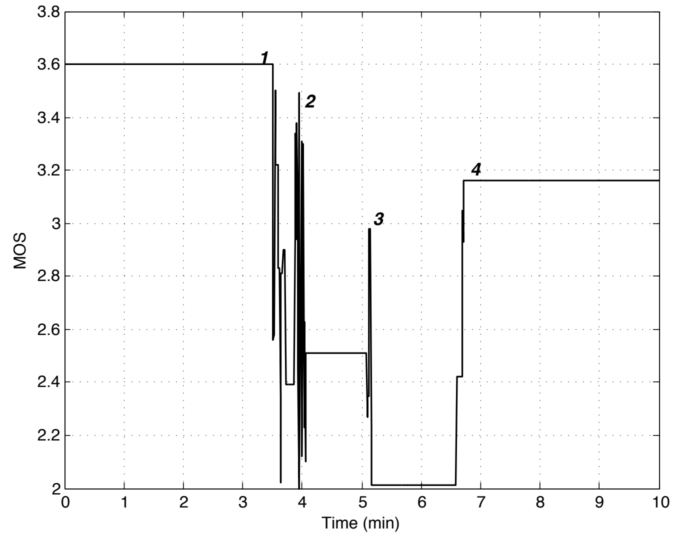

Fig. 4. RSS-based Scheme

terminal is performed at approximately the fourth minute by selecting a base station with the MOS value of 2.5. As the MOS value is lower, the user experiences poor call quality. After the fifth minute the mobile terminal again experiences poor RSS and performs a handover. This time the MOS value of the base station is slightly better than the previous one but not the best one to service the call.

It can be seen from Figure 3 and Figure 4 that MOS-based handover scheme provides better MOS value than the RSS-based scheme. Since MOS value is a QoS parameter which is calculated from the delay and packet loss, this indicates that the new MOS-based handover scheme is better than the RSS-based scheme in terms of maintaining quality of the call. The drawback of our MOS-based scheme is, the processing time at the RNC is more than the RSS-based scheme, as it involves the collection of delay and packet loss parameters, and then, calculation of the MOS for the candidate base station once the handover request is made.

2) Handover Dropping Probability (HDP): When a mobile terminal requests for a handover, the handover process is dropped if a handover request is not processed. The corresponding probability is called as handover dropping probability. The call still continues with the current attached base station. The dropping probability is given by $p(d) = x/y$, where x is number of unsuccessful handover and y is number of handover requests [22]. HDP is a QoS metrics and can be used as an performance indicator of a system. When a mobile terminal moves from one cell to another cell or from one network to another network, the call has to be transfered without dropping or degrading the quality. For a mobile terminal to maintain seamless connectivity, HDP plays a very important role.

For both MOS and RSS based schemes we calculated HDP for 100 users. Figure 5 shows handover dropping probability in MOS-based and RSS-based handover schemes. The HDP of

RSS-based handover is higher than the MOS-based handover schemes. The handover process fails for various reasons such as, no sufficient bandwidth, no enough wireless resources to provide strong signal strength to the mobile terminal receivers, more packet loss and delay in turn affecting MOS or could not support the application well. The handover dropping probability depends on the type of handover schemes. In case of RSS-based handover, Line of Sight (LOS) plays an important role in the success of the handover process. If a mobile terminal is not in the LOS it affects the RSS. When a user senses low signal strength and invoke a handover request, the possibility of high rate of rejection is true which in turn increases handover dropping probability. During the handover process, the packets are buffered at the mobile terminal and at the base station. Due to the limited buffer size or buffer overflow, the handover request is put in the queue and serviced according to the First-In-First-Out procedure. The handover is dropped if the waiting time expires. If the mobile terminal is moving in the heterogeneous network, for RSS-based handover the user equipment has to have multiple antennas to detect the signal strength from multiple networks. But in case of our handover scheme, it is not necessary for a mobile terminal to have multiple antennas since we use MIH protocol to detect the link layer state.

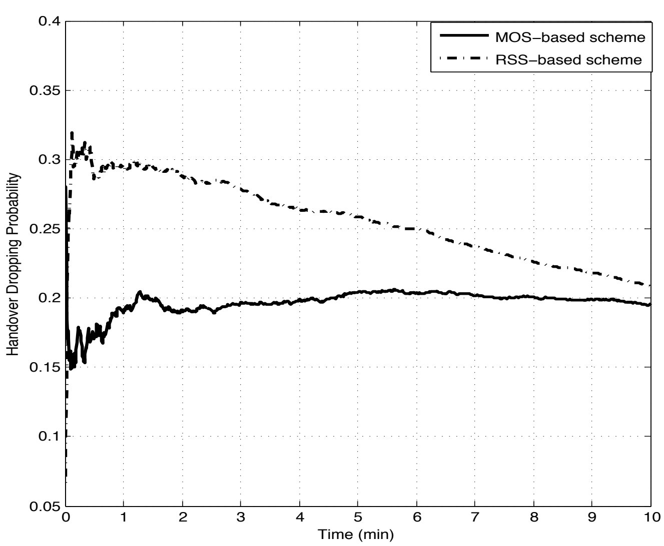

Fig. 5. Handover dropping probability

3) Call Dropping Probability (CDP): When a handover request cannot be processed and the call cannot be serviced either by the current base station or the candidate base station, the call is dropped and the probabilty is called call dropping probability. The call dropping probability is given by $p(c) = x/y$, where x is number of unsuccessful calls and y is total number of call requests.

CDP seems to be confused with HDP. These are two different parameters but are inter-related. When a mobile terminal requests for the handover, the request can be granted or denied. When the handover request is denied, the handover process is dropped. The corresponding probability is called handover dropping probability, the call still continues on the current base station. But in case of CDP, the call is dropped as it cannot be serviced by any of the base stations. CDP is a subset of HDP but not vice a versa. The call dropping rate is used as a indicator to identify network congestion or to realize the base station has a higher packet loss and delay. Call dropping is more serious than handover dropping. HDP and CDP can be caused by poor signal strength, scarce wireless resources, wireless transmission delay and wireless channel access delay. It increases as the congestion in the network increases or due to poor reception caused by the fading signal. The more the number of handovers per base station, the more will be the buffering of packets at the base station, which in turn increases the HDP and CDP.

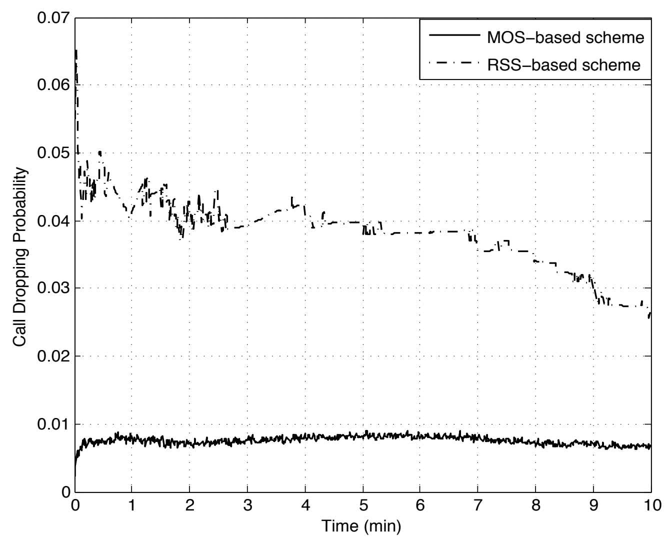

Fig. 6. Call dropping probability

For both MOS and RSS based scheme we calculated CDP for 100 users. Figure 6 shows call dropping probability in MOS-based and RSS-based handover schemes. RSS-based scheme has higher call dropping probability than the proposed MOS-based scheme. It can be seen from the Figure 5 that in RSS-based scheme more handover requests are dropped. Since HDP is higher and if mobile terminal cannot be serviced by current base station, the call is dropped as well. As seen from Figure 6, MOS-based handover has lower call dropping probability than the RSS-based scheme. When the current base station cannot service the call, the mobile terminal requests for a handover. When the mobile terminal makes a request for handover, base station has to minimize the number of dropping calls. The call is dropped if the mobile terminal cannot be serviced by candidate base station as well as current base station.

VII. CONCLUSION

Handover is necessary when a connection needs to be transferred between cells or networks for seamless connectivity and good quality of call. In this paper, we proposed a novel handover scheme using IEEE 802.21 standard, that enables a wireless access network to transfer the call between cells or networks, taking care of the quality of the call and load among all the available attachment points. We formulated the base station selection problem as an optimization problem with the objective to maximize the call quality, and presented a scheme to forward data packets to the most appropriate attachment point in order to maintain good call quality. We conducted extensive simulation using a scenario of urban network environment with VoIP call in WiMAX and UMTS integrated networks and analyzed critical QoS parameters like MOS, CDP and HDP. We compared our proposed scheme with the RSS-based handover scheme. Results show that our proposed scheme provides higher MOS values thus improving the perceived quality of the call and reduces the HDP and CDP. It is a QoS aware scheme which guarantees the call quality to the user. The proposed scheme is energy efficient as it does not require to scan the network frequently. Future work includes evaluation of the energy efficiency of our scheme.

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