International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 1 Issue 6, August - 2012

Smooth Adaptive Soft Handover Algorithm

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Abstract

Inter-network mobility is achieved by allowing a mobile node to change its point of attachment to the network while preserving connectivity to its corresponding nodes. Handover management becomes an increasingly important component of the emergent mobile Internet by maintaining mobile user's data sessions alive in the presence of user mobility. The great impact handover has on Quality of Service makes it a crucial factor in maintaining mobile user's Quality of Experience at a high level. This paper proposes the Smooth Adaptive Soft- Handover Algorithm (SASHA) which increases the quality of the multimedia delivery process when performing handover in heterogeneous wireless environment by gracefully transferring the load from one connection to the other.

1. Introduction

Mobile networking has encountered a rapid growth with the latest advancements in wireless technologies and increase in the number of mobile computing devices. At the same time multimedia applications like IPTV, video-on-demand, distance learning and teleconferencing are some examples of multimedia applications that are attracting increasing interest among mobile device users. As the Internet was originally designed for communication between fixed devices, mobility management should be additionally provided to accommodate mobile devices. Multimedia streaming to mobile users over wireless networks involves difficult challenges in terms of mobility support due to the high sensitivity of multimedia quality to network conditions.
Different solutions such as Mobile IP, Mobile SCTP, Mobile DCCP, etc. were developed to support handover at different layers. [1]

2. Literature Review

Various handover mechanisms were proposed in the literature at almost all the layers of the network protocol stack. The most popular are Mobile IP, at the network layer, Mobile SCTP and DCCP at the transport layer and Mobile SIP at session layer. For network capacity estimations and network selection several solutions were proposed in the literature. Several categories of network selection algorithms can be identified including function based strategies, user centric strategies, multiple attribute and context aware decision strategies, as well as strategies based on fuzzy logic and neural network. All the above presented solutions are designed for single network selection and not for traffic distribution over multiple simultaneous networks. Rate control is widely investigated and implemented at various layers of the network protocol stack. Among the most popular are Transmission Control Protocol's (TCP) congestion control mechanisms and the TCP-Friendly Rate Control (TFRC) as well as the application layer adaptive multimedia streaming techniques such as QOAS. Traffic distribution over multiple networks has also been widely investigated in the literature such as the one proposed as DRA. [2]

Existing Systems:

The transmission of data content over heterogeneous wireless networks to mobile devices involves significant technical challenges related to mobility management and quality of service provisioning. The existing solutions do not consider quality of service as a decision making parameter for mobility management in general and handover management in particular. [2] Most of these proposed solutions directly change the whole data flow from one network to another as the mobile nodes roams through different networks coverage area. [3]

3. Smooth Adaptive Soft Handover Algorithm

However the existing system's main drawback is the lack of a quality oriented approach, which would combine handover, network selection and load balancing techniques in order to maximize user perceived quality by efficiently exploiting all the communication resources currently available. This paper proposes the Smooth Adaptive Soft Handover Algorithm (SASHA), as an application-layer quality-aware approach to handover, based on load balancing among different networks. This proposed handover algorithm exploits both the old and the new connections to transfer multimedia data when the user is crossing the two networks overlapping area. In this context SASHA transfers gracefully multimedia streaming process from the old fading connection to the new improving one. This operation is performed efficiently without data duplication. Three aspects of wireless data communications are tightly related to the scheme investigated in this paper: handover management, network selection and traffic rate allocation.

The objective is to develop a simulation system for handover management system using smooth adaptive soft handover algorithm (SASHA) for data transmission in MANET.

4. System Architecture

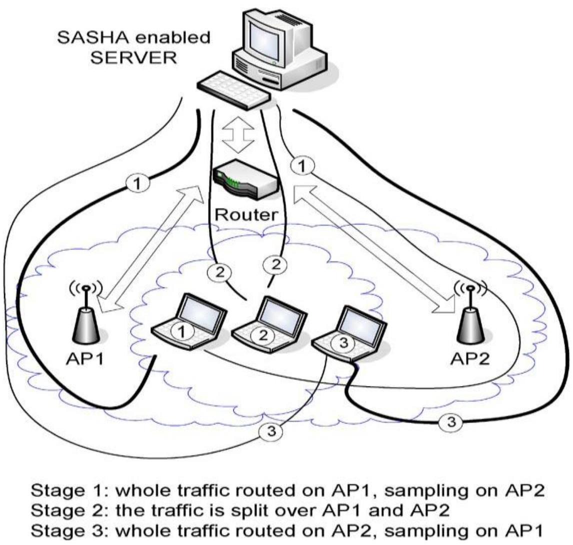

Figure 1: Handover operation using SASHA

Figure 1 presents schematically a horizontal handover performed using SASHA involving two networks using infrastructure modes and having AP1 and AP2 as access points (access routers). The assumption that the mobile node is equipped with two network interfaces is made. The example refers to horizontal handover but the same technique can be used for vertical handover considering an all-IP network environment. Although the handover process is continuous, for simplicity three different stages will be identified and presented.

In stage 1 the MN is currently communicating entirely over AP1 and enters the overlapping area. When the link via AP2 becomes available, MN opens a new connection to the server which sends a low bit rate sampling stream over the new AP2 path for QMS computation. As QMS metric is evaluated for the two connections and mainly due to the high distance to AP2, QMS2 is very much lower than QMS1. In stage 2 MN moves towards AP2 determining QMS decrease for the AP1 path and QMS increase for the AP2 connection. As QMS values change, SASHA server starts gradually to increase the share of multimedia content delivered over the new AP2 connection and decrease the load of the AP1 connection. In stage 3 MN is about to leave the overlapping area and enter exclusively in the AP2 coverage area. In this situation, as QMS value for the AP1 link decreases significantly, whereas the QMS value for the AP2 connection becomes very high, all multimedia traffic will be routed over AP2 connection. However AP1 connection is still sampled while still available in order to compute QMS values, allowing the handover process to be reversed, if MN moves back towards AP1. [1]

5. Mathematical Model

M3S make use of a comprehensive Quality of Multimedia Streaming (QMS) function for decision making, which combines QoS, QoE, cost, energy and user preference components. M3S is designed as an application layer module used by the multimedia applications to efficiently deliver high quality multimedia content to mobile users. Multimedia content deliveries over various communication links which differ in terms of technology, load, cost and even protocol influence differently multimedia quality. [3] Therefore the Quality of Multimedia Streaming metric is used to describe and quantify the effect all these factors have on the multimedia delivery quality. QMS is represented by the function from equation (1) and is dependent on the characteristics of the connection i. [1]

$$QMS^i = w_1 * QoS^i_{grade} + w_2 * QoE^i_{grade} + w_3 * cost^i_{grade} + w_4 * PEff^i_{grade} + w_5 * UPref^i_{grade} \quad \ldots\ldots (1)$$

For maximum efficiency and flexibility weights are associated with each component. These weights are set based on user preferences and application requirements. Weights normalization is required, so the condition from (2) has to be respected.

$$\sum_1^5 wi \quad \ldots\ldots (2)$$

6. ALGORITHM: SASHA Rate Adaptation

Step 1: START

Step 2: Input: Relevant data to compute QMS for each path such as Number of received packets-$recv_i$; lost packets-$loss_i$;delay-$delay_i$; jitter-$jitter_i$; cost per Mbp-$cost_i$ etc

Step 3: Procedure: Update_Rate

Step 4: Compute QMS_i

Step 5: If QMS variation > Threshold then

Step 6: Select the paths to be used, P_j

Step 7: Compute rate share RS_j for each path P_j

Step 8: R_j=TargetBit rate*RS_j

Step 9: STOP

Output R_i-sending rate for path i

This presents the pseudo-code of a simplified version of SASHA rate adaptation algorithm. Rate update (Update_Rate) is performed each time QMS related feedback is received from the client, or new information is harvested from the lower network layers. If the variation in QMS is significant according to the required algorithm sensitivity (a threshold value was introduced), the rate adaptation procedure is triggered. The first step consists of communication channel selection. Based on QMS values the first best channels are selected which gather enough efficient traffic capacity to deliver high quality multimedia content at the target bit rate. In the next step the rate share is computed for each communication channel according to the QMS scores and application requirements. The QMS scores are expressed on a 100 point scale and represent the estimated share (expressed in percentage) of the total streaming rate that a certain connection can transport at high quality. The rate share (RS) associated with a connection represents the fraction of the total streaming rate which can be transported at high quality over that connection and is calculated according to the connection's QMS score. The actual sending rate (R) expressed in Mbps is computed from the target bit rate and the previously computed RS parameter. The last step distributes the traffic load according to the rate shares computed in the preview step.

The QMS scores are computed by the server side module while the QMS parameters are harvested by the client side module. Consequently the proposed solution involves a certain network overhead determined by QMS feedback sent by the client to the server. The QoS and QoE parameters are sent more frequent while the other QMS parameters (i.e. user preferences, cost, etc.). Solutions like MIP and Mobile DCCP present less network overhead as the decision is made by the mobile device (client) and only a location update is required. However if these mobility management solutions are used in conjunction with a feedback-based adaptive multimedia streaming scheme, when using M3S there is a significant advantage of sending the feedback information only once and therefore reducing the overall overhead. [3].

7. NCBP and HCDP

User's QoS requirements can be quantitatively expressed in terms of probabilistic connection-level QoS parameters (related to connection establishment and management) such as new call blocking probability (NCBP) and handoff call dropping probability (HCDP). A new call is initiated when a user requests a new connection, while a handoff call occurs when an active user moves from one cell to another neighboring cell. Thus, the NCBP is the probability of a new arriving call being rejected while the HCDP is the probability that an accepted call is terminated before the completion of its service, i.e., the probability that a handoff attempt fails. Providing multimedia services with QoS guarantees in MWNs presents great challenges due to 1) the limited bandwidth; and 2) the high rate of handoff events. While minimizing the HCDP is very desirable from the user's point of view, this often comes at the expense of the resource (e.g., bandwidth) utilization, which is very undesirable from the service provider's point of view. When the system is underutilized, all arriving calls (new and handoff) are admitted and assigns the highest bandwidth level that results in increasing of the bandwidth utilization. The objective of this framework is reducing the NCBP and the HCDP. [6]

Adaptive Bandwidth Allocation:

It contains two main procedures: reduction and expansion. The reduction procedure is activated when an accepted arriving call (new or handoff) arrives to an overloaded cell. On the other hand, the expansion procedure is activated when there is an outgoing handoff call or a call completion in the given cell. The process flows for both procedures, reduction and expansion, are given in Fig.2a and Fig.2b.
Whenever system accepts an arrival call (new or handoff), the system attempts to allocate maximum bandwidth (b_n) for this call. Thus, if the available bandwidth is larger than or equal to $b_{requected,}$ the arrival call will be assigned a bandwidth between $b_{requected}$ and b_n. Otherwise; a reduction procedure is invoked to reduce the bandwidth of some ongoing calls in the cell as follows. Calls with the largest assigned bandwidth greater than $b_{requected}$ are reduced to have lower bandwidth not less than $b_{requected.}$ If the saved bandwidth is larger than or equal to $b_{requected,}$ the arrival call will assigned a bandwidth between $b_{requected}$ and b_n . Otherwise, we do further bandwidth reduction to accommodate the call. Thus, calls with the largest assigned bandwidth in the cell are reduced to have lower bandwidth not less than the minimum bandwidth (i.e., b_n). If the saved bandwidth is larger than or

equal to $b_{requected}$, the arrival call will be assigned a bandwidth between $b_{requected}$ and b_n.Otherwise, we just assign the saved bandwidth to the call (minimum is b_n). If all above tests fail, then block/drop arrival call. As a call leaves the cell, whether outgoing handoff call or a call completion, the total available bandwidth increases. The system will invoke the expansion procedure to increase the bandwidth for one or more of the degraded calls to $b_{requected}$', starting from most degraded calls in the cell. Expansion procedure stops when there is no available bandwidth or every call in the cell has a bandwidth larger than or equal to $b_{requected}$.[5]

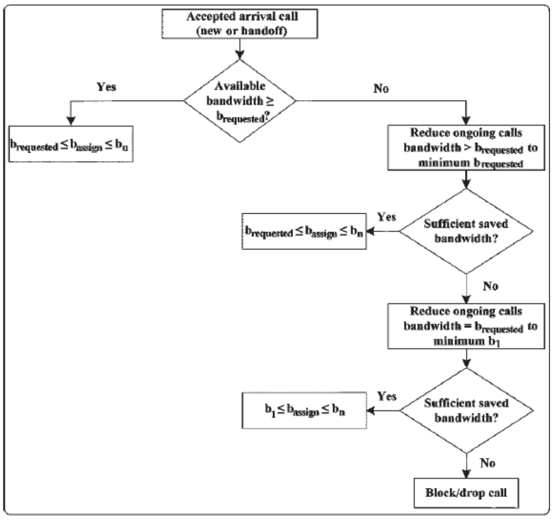

Figure 2 a) Bandwidth allocation when an accepted arriving call (new or handoff) arrives to an overloaded cell

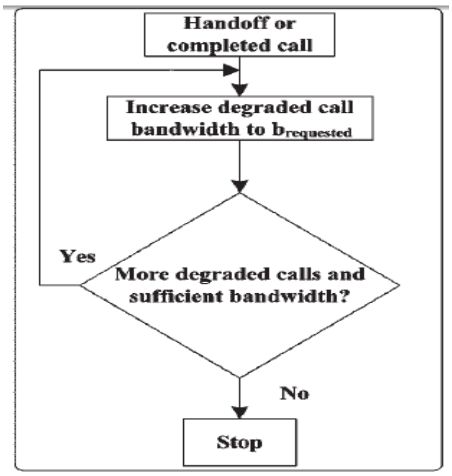

Figure 2 b) Bandwidth allocations when there is an outgoing handoff call or a call completion in the given cell.

7.1 Adaptability Constraints

Adaptability describes how many times we can adapt the system before we hit the state where we cannot adapt the system anymore. The number that describes the adaptability of the system is called the adaptability ratio or alpha(α). The adaptability ratio measure simply depends on the number of calls in that level (i.e.,b_I). It is equal to one minus the sum of the allocated bandwidth for all levels divided by the total bandwidth in the cell. The total allocated bandwidth for all active users for all levels can be computed from

$$BW_{allocated}=\sum xibi\ldots\ldots\ldots\ldots\ldots\ldots\text{for } i=1 \text{ to } n$$

Where x_i is the number of current users that are allocated level i and b_i is the bandwidth allocated for level i users. The adaptability ratio can be then calculated as follows:

$$\alpha=1- BW_{allocated}/C$$

Notice that the adaptability ratio depends only on the number of calls of that level in the cell at the moment since all of the other variables in the above two presumably constants. The extra step is to calculate the adaptability ratio during the bandwidth adaptation process and triggering algorithm for new calls if the adaptability ratio of the system is larger than a predefined value, beta (β). Figure shows the adaptability framework.[5]

8. Experimental setup

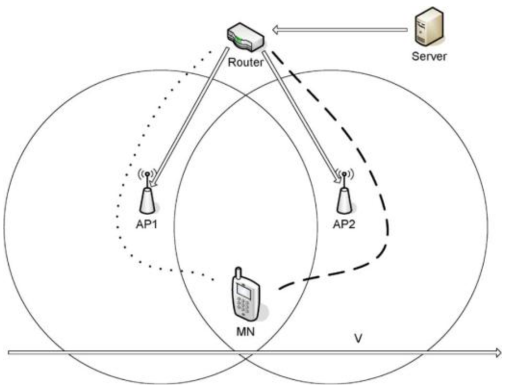

Figure 3: Simulation Scenario

The behavior and the performance of the proposed mobility solution is evaluated based on simulations conducted using the NS-2 Network Simulator (v2.29). With the help of network simulator software mobile nodes are created. There are 16 mobile nodes. They are created in such a way that there are some senders and some receivers; the data transmission is performed simultaneously among them. Among sixteen mobile nodes, the two nodes are serves as BS (Base Station). The two BSs were positioned close enough to each other to provide a coverage overlapping area. When simulation occurs, after some time the nodes will start changing their positions, and three nodes from them are considered as new nodes that will be entering into the existing simulation. Now, the data can be transfer to these nodes. DSDV routing is performed between nodes. With the help of number of total request calls and number of handoff request calls, calculate the call arrival rate and bandwidth utilization of the system at the current time. Handoff and mobility scenario is applied in such a way that new call blocking and handoff call dropping probabilities are reduced. Handoff and mobility management is performed with the variation in network parameter like different beta values, variation in total new call request and handoff call request. The system will perform efficiently though the mobile nodes are changing their positions. SASHA is employed to handle the mobility management under load balancing situation.

9. Simulation results and Analysis

Analysis of Bandwidth utilization with HCDP

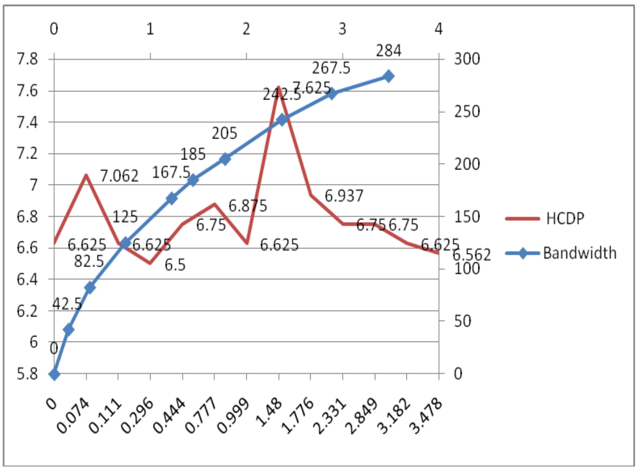

Figure 4: Bandwidth and HCDP

As the above figure shows the maximum bandwidth for the peak value of HCDP is 267.5 Mbps and lowest value of HCDP is 90Mbps.This indicates 267.5-90/available bandwidth (284) Mbps. This is equal to 62.5% of available bandwidth is utilized.

Analysis of Bandwidth utilization with NCBP

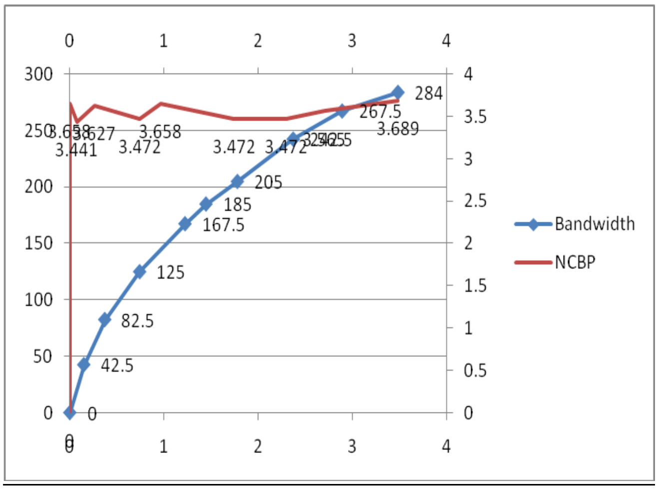

Figure 5: Bandwidth and NCBP

The maximum call blocking probability is 3.689 and minimum is 3.441 for which corresponding bandwidth is 284 and 26.This indicates that 284-26/284 equals to 90.8% utilization is observed.

Analysis of New Call Blocking Probability (NCBP) with variations on total calls and beta values

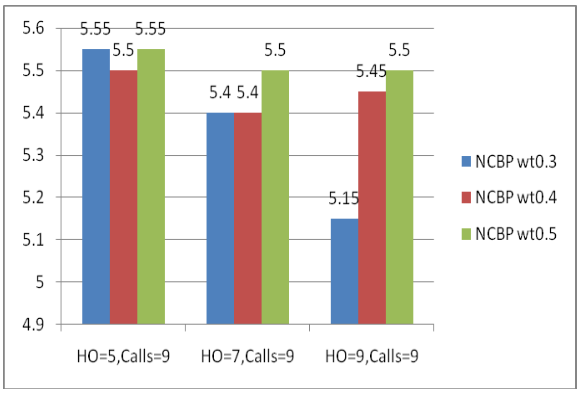

Figure 6: Variation in parameters for NCBP

These figures shows the New Call Blocking Probability for the Beta Values 0.3, 0.4 and 0.5

The set of values are plotted for number of calls 9, number of calls 11, and number of calls 13.

The set of values for Handoff are considered as 5, 7 and 9.

It is observed that though total calls increases from 9 to 13 the probability decreases from 7% to 2.7%

As New Call(Connection) increases from 9 to 13 the New Call Blocking Probability decreases from 5.55 to 4.4 and then to 3.7 for given set of values.

For the same Handoff 5 and total calls 9 and 11, 20.72% decrease in the probability observed.

For total calls 11 and total calls 13, 15% decrease in probability observed resulting in New Call Blocking Probability reduction.

Above results indicate that the SASHA Algorithm takes care of variation in the system that is though New call increases as well as Handoff increases New Call Blocking probability decreases.

Analysis of Handoff Call Dropping Probability (HCDP) with variations in number of handoffs and beta values:

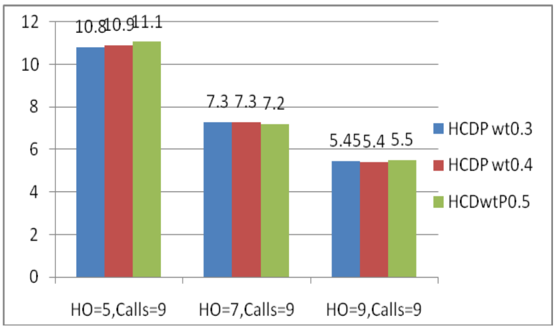

Figure 7:Variation in parameters for HCDP

These figures show Handoff Call Dropping Probability for the Beta values of 0.3, 0.4, and 0.5.The set of values are plotted for number of calls 9, number of calls 11 and number of calls 13.The set of values for Handoff are considered as 5, 7 and 9.

With respect to figure 7, indicates the effect of increase in the Handoff calls for the same total call request 9. When Handoff call increases from 5 to 7, the Handoff Call Dropping Probability decreases to 35%. Similarly when Handoff increases from 7 to 9 the Handoff Dropping probability decreases to 25%.

When the effect of increase in the Handoff for same total calls 11:

When Handoff call increases from 5 to 7 the Handoff Call Dropping Probability to 37%. Similarly when Handoff increases from 7 to 9 the Handoff dropping probability decreases to 25%.

When the effect of increase in the Handoff for same total calls request of 13:

When Handoff increases from 5 to 7 the Handoff dropping probability decreases to 36%. Similarly when Handoff increases from 7 to 9 the Handoff dropping probability decreases to 26%. As above result indicates that SASHA algorithm takes care of variation in the system that is though number of Handoff increases and Simultaneously New call request increase the Handoff dropping probability decreases. It shows the efficiency of the system for variable set of parameters.

10. CONCLUSION

Efficiently distributing the traffic over multiple wireless networks improves the user perceived quality in highly loaded network conditions. Including the monetary cost and energy consumption components improves the performance of SASHA while maintaining the QoS and QoE at high levels. As delivering multimedia content to mobile devices over IP networks becomes increasingly popular, SASHA gives a quality-oriented mobility solution. This solution aims at maximizing the end-users perceive quality when streaming multimedia content by efficiently using all the communication resources available. Smooth Adaptive Soft Handover Algorithm (SASHA) gracefully and dynamically distribute the load over the available communication channels based on their estimated contribution in order to deliver high Quality of Service.

When Bandwidth is compared with HCDP 62.5% of available bandwidth is utilized. Similarly when it is compared with NCBP 90.8% utilization is observed. As the Throughput increases from 0 to 86.62 Mbps the packet loss decreases by 76% and delay decreases by 60%. Analysis of New Call Blocking Probability (NCBP) with variations on total calls and beta values shows that, though total calls increases from 9 to 13 the probability decreases from 7% to 2.7%. Results indicate that the SASHA Algorithm takes care of variation in the system that is though new call increases as well as Handoff increases New Call Blocking probability decreases. Analysis of Handoff Call Dropping Probability (HCDP) with variations in number of handoffs and beta values describes that when Handoff increases from 5 to 7 the Handoff dropping probability decreases to 36%. Similarly when Handoff increases from 7 to 9 the Handoff dropping probability decreases to 26%.Thus as above result indicates that SASHA algorithm takes care of variation in the system that is though number of Handoff increases and Simultaneously New call request increase the Handoff dropping probability decreases. It shows the efficiency of the system for variable set of parameters

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