Implementation of Saurp for Mobile Networks

Krishna Sowmya Chinta [#1] SD Afzal Ahmed [#2] P. Babu [#3]

[1]Student, Department of Computer Science and Engineering
QUBA college of Engineering and Technology, Nellore, A.P., INDIA.
[2] Associate Professor, Department of Computer Science and Engineering
QUBA college of Engineering and Technology, Nellore, A.P., INDIA.
[3] Head of the Department, Department of Computer Science and Engineering
QUBA college of Engineering and Technology, Nellore, A.P., INDIA

Abstract—Delay Tolerant Networks (DTNs) that are possibly composed of a vast number of devices such as smart phones of diverse capacities in terms of energy resources and buffer spaces. We introduces protocol which is a novel multi-copy routing protocol called Self Adaptive Utility-based Routing Protocol (SAURP) for DTNs. SAURP has the ability to identify potential opportunities of forwarding messages to their destinations via a novel utility function based mechanism, where a suite of environment parameters, such as wireless channel condition, nodal buffer occupancy, and encounter statistics, are jointly considered. Taking a considerably small number of transmissions, it can reroute messages around nodes experiencing high buffer occupancy, wireless interference, and congestion. Thus SAURP utility function is proved to achieve optimal performance and it is also further analyzed via a stochastic modeling approach. To verify the developed analytical model, extensive simulations are conducted and are compared the proposed SAURP and shown that SAURP can perform all the counterpart multi-copy encounter-based routing protocols. These simulations are associated with the number of recently reported encounter-based routing approaches in terms of delivery ratio, delivery delay, and the number of transmissions required for each message delivery.

Index Terms—Encounter-based Routing, DTN.

I. INTRODUCTION

Delay Tolerant Network (DTN) [1] is characterized by the lack of end-to-end paths for a given node pair for extended periods, which poses a completely different design scenario from that for conventional mobile adhoc networks (MANETs) [8]. Due to the intermittent connections in DTNs, a node is allowed to buffer a message and wait until the next hop node is found to continue storing and carrying the message. Such a process is repeated until the message reaches its destination. This model of routing is significantly different from that employed in the MANETs. DTN routing is usually referred to as encounter-based, store-carry-forward, or mobility-assisted routing, due to the fact that nodal mobility serves as a significant factor for the forwarding decision of each message.

Depending on the number of copies of a message that may coexist in the network, two major categories of encounter-based routing schemes are defined: single-copy and multi-copy. With the single-copy schemes [4], no more than a single copy of a message can be carried by any node at any instance. On the other hand, multiple-copy (or multi-copy) routing schemes allow the networks to have multiple copies of the same message that can be routed independently and in parallel so as to increase robustness and performance.

Although improved in terms of performance, the previously reported multi-copy schemes are subject to the following problems and implementation difficulties. First, these schemes inevitably take a large number of transmissions, energy consumption, and a vast amount of transmission bandwidth and nodal memory space, which could easily exhaust the network resource. Second, they suffer from contention in case of high traffic loads, when packet drops could result in a significant degradation of performance and scalability. Note that the future DTNs are expected to operate in an environment with a large number of miniature hand-held devices such as smart phones, tablet computers, personal digital assistants (PDAs), and mobile sensors. In such a scenario, it may no longer be the case that nodal contact frequency serves as the only dominant factor for the message delivery performance as that assumed by most existing DTN literature. Therefore, limitations on power consumption, buffer spaces, and user preferences should be jointly considered in the message forwarding process.

To cope with the above-mentioned deficiency, a family of multi-copy schemes called utility-based controlled flooding [10], [12], [9], [7] has been proposed. The class of schemes generates only a small number of copies to ensure that the network is not overloaded with the launched messages. Although being able to effectively reduce the message delivery delay and the number of transmissions, most of the utility-based controlled flooding routing schemes in literature assume that each node has sufficient resources for message buffering and forwarding. None of them, to our best knowledge, has sufficiently investigated how the protocol should take advantage of dynamic network status to improve the performance, such as packet collision statistics, wireless link conditions, nodal buffer occupancy, and battery status. Note that the nodal buffer status could

serve as an indicator how much the opportunity cost is by accepting a forwarded message; while the channel condition is an indicator how likely the contact could be an eligible one; or in other words, how likely a message can be successfully forwarded during the contact. They are obviously essential parameters to be considered in the utility function.

With this in mind, we introduce a novel DTN routing protocol, called Self Adaptive Utility-based Routing Protocol (SAURP), which aims to overcome the shortcomings of the previously reported multi-copy schemes. Our goal is to achieve a superb applicability to the DTN scenario with densely distributed hand-held devices. The main feature of SAURP is the strong capability in adaptation to the fluctuation of network status, traffic patterns/characteristics, user encounter behaviors, and user resource availability, so as to improve network performance in terms of message delivery ratio, message delivery delay, and number of transmissions.

The contributions of the paper are as following
- We develop a novel DTN routing scheme which incorporates with some parameters that have not been jointly considered in the literature. The parameters include link quality/availability and buffer occupancy statistics, which are obtained by sampling the channels and buffer space during each contact with another node.
- We introduce a novel transitivity update rule, which can perfectly match with the proposed routing model and the required design premises.
- We introduce a novel adaptive time-window update strategy for maintaining the quality metric function at each node, aiming at an efficient and optimal decision making process for each active data message.
- An analytical model is developed for the proposed SAURP, and its correctness is further verified. We show via extensive simulations that the proposed SAURP can achieve significant performance gain over the previously reported counterparts under the considered scenarios.

The rest of the paper is organized as follows. Section II provides a review of the related work. Section III describes the proposed SAURP in detail. Then, Section IV analyzes the proposed SAURP by coming up with an analytical model for the delivery ratio and end-to-end message delivery delay. Section V provides the simulation results and comparisons with the counterparts. In Section VI, we conclude the paper.

II. RELATED WORK

Most (if not all) previously reported encounter-based routing schemes have focused on nodal mobility, which has been extensively exploited as the dominant factor in the message forwarding decision. Those schemes contributed in the context of introducing new interpretations of the observed node mobility in the per-node utility function. Spyropoulos et al. in [5], [7] developed routing strategies that use different utility routing metrics based on nodal mobility statistics, namely Most Mobile First (MMF), Most Social First (MSF), and Last Seen First (LSF). Tan et al. in [17] introduced a routing strategy based on calculating the expected end-to-end path length as a metric in forwarding messages mainly based on the reciprocal of the encountering

probability. It is defined as the expectation of message transmission latency through multi-hop relays. Spyropoulos et al. in [5] proposed routing scheme called Spray and Focus, which is characterized by addressing an upper bound on the number of message copies (denoted as L). In specific, a message source starts with L copy tokens. When it encounters another node B currently without any copy of the message, it shares the message delivery responsibility with B by transferring L=2 of its current tokens to B while keeping the other half for itself. When it has only one copy left, it switches to a utility forwarding mechanism based on the time elapsed since the last contact. This scheme has proven to significantly reduce the required number of transmissions, while achieving a competitive delay with respect to network contentions such as buffers space and bandwidth. Mosli et al. in[11] introduced a DTN routing scheme using utility functions calculated from an evaluation of context information. The derived cost function is used as an assigned weight for each node that quantifies its suitability to deliver messages to an encountered node regarding a given destination. Other schemes are based on content-based network service [19] as a novel style of communication that associates source and destination pairs based on actual content and interests, rather than by letting the source to specify the destination.

Although previously reported studies such as [5], [6], [2], [9], [31], [13] have made great efforts in improving the DTN routing techniques, they are subject to various limitations in the utility function update processes. The schemes such as [2], [18] that take the number of encounters as the main factor in the message forwarding decision, may suffer from multiple falsely detected contacts. Further, a permanent or quasi-permanent neighbor will cause the utility function calculation invalid in message forwarding.

Although the above mentioned schemes can capture the mobility properties in order to come up with efficient forwarding strategy, they may not be able to acquire accurate knowledge about network dynamics and unpredicted contacts. More importantly, the channel capacity and buffer occupancy status have never been jointly considered in the derivation of utility functions for hop-by-hop message forwarding. It is clear that these two factors could only be overlooked/ignored when the encounter frequency is low since the routing protocol performance is dominated by node mobility, while the network resource availability does not play an important role. However, in the scenario that the nodal encounter frequency is large and each node has many choices for message forwarding in a short time, the network resource availability is envisioned to serve as a critical factor for performance improvement and should be utilized in the derivation of utility functions.

Motivated by the above observations, this paper investigates encounter-based routing that jointly considers nodal contact statistics and network status including wireless channel condition and buffer occupancy. Our goal is to reduce the delivery delay and the number of transmissions under stringent buffer space and link capacity constraints. This is a desired feature of a DTN especially in the scenario where each mobile node is hand-held device with limited resources.

III. SELF ADAPTIVE UTILITY-BASED ROUTING PROTOCOL (SAURP)

The proposed SAURP is characterized by the ability of adapting itself to the observed network behaviors, which is made possible by employing an efficient time-window based update mechanism for some network status parameters at each node. We use time-window based update strategy because it is simple in implementation and robust against parameter fluctuation. Note that the network conditions could change very fast and make a completely event-driven model unstable.

A. Contact Statistics (CS)

To compromise between the network state adaptability and computation complexity, each node continuously updates the network status over a fixed time window. The maintained network states are referred to as Contact Statistics (CS), which include nodal contact durations, channel conditions, and buffer occupancy state, and are fed into UCUM at the end of each time window. The CS collection process is described as follows.

Let two nodes A and B be in the transmission range of each other, and each broadcasts a pilot signal per k time units in order to look for its neighbors within its transmission range. Let $T_{(A,B)}$, T_{free}, and T_{busy} represent the total contact time, the amount of time the channel is free and the buffer is not full, and the amount of time the channel is busy or the buffer is full, respectively, at node A or B during time window $W^{(i)}$. Thus, the total duration of time in which node A and B can exchange information is calculated as:

$$T_{free} = T_{(A,B)} -- T_{busy} \qquad (1)$$

Note that the total contact time could be accumulated over multiple contacts between A and B during $W^{(i)}$.

B. Utility-function Calculation and Update Module (UCUM)

UCUM is applied at the end of each time window and is used to calculate the currently observed utility that will be further used in the next time window. The two inputs to UCUM in time window $W^{(i)}$ are: (i) the predicted inter-contact time ($\Delta T_{(i)}$), which is calculated according to the previous time-window utility (i.e., $\Delta T^{(i)}$), as well as an update process via the transitivity property update (introduced in subsection 3.3), and (ii) the observed inter-encounter time obtained from the current $CS^{(i)}$ (denoted as $\Delta T^{(i)}_{cs}$).

1) Calculation of Inter-encounter Time ($\Delta T^{(i)}$)
An eligible contact of two nodes occurs if the duration of the contact can support a complete transfer of at least a single message between the two nodes. Thus, in the event that node A encounters B for a total time duration T_{free} during time window $W^{(i)}$, the number of eligible contacts in the time window is determined by:

$$n^{(i)}_c = \left\lfloor \frac{Tfree}{Tp} \right\rfloor \qquad (2)$$

where T_p is the least time duration required to transmit a single message. Let $\Delta T^{(i)}_{cs(A,B)}$ denotes the average inter-encounter time duration of node A and B in time $W^{(i)}$. Obviously, $\Delta T^{(i)}_{(A,B)} = \Delta T^{(i)}_{(B,A)}$. We have the following expression for $\Delta T^{(i)}_{cs(A,B)}$:

$$\Delta T^{(i)}_{cs(A,B)} = \frac{W(i)}{n(i)c} \qquad (3)$$

$\Delta T^{(i)}_{cs(A,B)}$ describes how often the two nodes encounter each other per unit of time (or, the encounter frequency) during time window $W^{(i)}$ considering the event the channel is busy or the buffer is full.

2) Time-window Transfer Update
Another important function provided in UCUM is for the smooth transfer of the parameters between consecutive time windows. As discussed earlier, the connectivity between any two nodes is measured according to the amount of inter-encounter time during $W^{(i)}$, which is mainly based on the number of contacts (i.e., n_c) and the contact time (i.e., T_{free}). Hence, our scheme determines the next time window parameter using two parts: one is the current time window observed statistics (i.e., $\Delta T^{(i)}_{cs}$), and the other is from the previous time window parameters (i.e., $\Delta T^{(i)}$), in order to achieve a smooth transfer of parameter evolution. The following equation shows the derivation of $\Delta T^{(i+1)}$ in our scheme.

$$\Delta T^{(i+1)} = \gamma \Delta T^{(i)}_{cs} + (1 - \gamma) \Delta T^{(i)} \qquad (4)$$

If $\Delta T^{(i)}_{cs} > W$, which happens if $n^{(i)}_c = 0$, then $\Delta T^{(i+1)} = \frac{2W}{nc(i-1)}$. This case represents a worst case scenario, i.e. unstable node behavior, or low quality of node mobility. Hence, the $\Delta T^{(i+1)}$ value should be low.

$\Delta T^{(i+1)}$ represents the routing metric (utility) value that is used as input to the next time window.

C. The Transitivity Update Module (TUM)

When two nodes are within transmission range of each other, they exchange utility vectors with respect to the message destination, based on which the custodian node decides whether or not each message should be forwarded to the encountered node. With a newly received utility vector, transitivity update [2] is initiated. We propose a novel adaptive transitivity update rule, which is different from the previously reported transitivity update rules [2],[5]. The proposed transitivity update rule is characterized as follows: (1) it is adaptively modified according to a weighting factor $\propto$, which is in turn based on the ratio of $\Delta T^{(i)}$ of the two encountered nodes regarding the destination rather than using a scaling constant. (2) It can quantify the uncertainty regarding the position of the destination by only considering the nodes that can effectively enhance the accuracy of the utility function.

The transitivity property is based on the observation that if node A frequently encounters node B and B frequently encounters node D, then A has a good chance to forward messages to D through B. Such a relation is implemented in the proposed SAURP using the following update strategy:

$$\Delta T^{(i)}_{(A,D)new} = \propto \Delta T^{(i)}_{(A,D)} + (1 - \propto)(\Delta T^{(i)}_{(A,B)} + \Delta T^{(i)}_{(B,D)}) \quad (6)$$

where $\propto$ is a weighting factor that must be less than 1 to be valid:

$$\propto = \frac{\Delta T(i)(A,B) + \Delta T(i)(A,B)}{\Delta T(i)(A,D)}, \Delta T^{(i)}_{(A,D)} > \Delta T^{(i)}_{(A,B)} + \Delta T^{(i)}_{(B,D)} \quad (7)$$

This rule is applied after nodes finish exchange messages.

D. The Forwarding Strategy Module (FSM)

The decision of message forwarding in SAURP is mainly based on the utility function value of the encountered node regarding the destination, and the number of message copy tokens. If more than one message copies are currently carried, the weighted copy rule is applied; otherwise the forwarding rule is applied.

1) Weighted Copy Rule

The source of a message initially starts with L copies. In the event that any node A that has $n > 1$ message copy tokens and encounters another node B with no copies with $\Delta T^{(i)}_{(B,D)} < \Delta T^{(i)}_{(A,D)}$, node A hands over some of the message copy tokens to node B and keeps the rest for itself according to the following formula:

$$N_B = \left\lfloor N_A \frac{\Delta T(i)(A,D)}{\Delta T(i)(B,D) + \Delta T(i)(A,D)} \right\rfloor \qquad (8)$$

where N_A is the number of message tokens that node A has, $\Delta T^{(i)}_{(B,D)}$ is the inter-encounter time between node B and node D, and $\Delta T^{(i)}_{(A,D)}$ is the inter-encounter time between nodes A and D. This formula guarantees that the largest number of message copies is spread to relay nodes that have better information about the destination node.

2) The Forwarding Rule
- If the destination node is one hop away from an encountered node, the custodian node hands over the message to the encountered node and completes the message delivery.
- If the inter-encounter time value of the encountered node relative to that of the destination node is less than that of the custodian node by a threshold value, $4T_{th}$, a custodian node hands over the message to the encountered node.

The complete mechanism of the forwarding strategy in SAURP is summarized as shown in Algorithm 1.

_____

_

Algorithm 1 The forwarding strategy of SAURP

On contact between node A and B
Exchange summary vectors
for every message M at buffer of custodian node *A do*
    if destination node D in transmission range of B
    then
        A forwards message copy to *B*
    end if
    else if $\Delta T^{(i)}_{(A,D)} > \Delta T^{(i)}_{(B,D)}$ *do*
        if message tokens >1 *then*
            apply weighted copy rule
        end if
        else if $4T^{(i)}_{(A,D)} > \Delta T^{(i)}_{(B,D)} + \Delta T_{th}$ *then*
            A forwards message to *B*
        end else if
    end else if
end   *for*
_____

IV. ANALYTICAL MODEL OF SAURP

In this section a statistical analysis is conducted to evaluate the performance of SAURP. Without loss of generality, Community-Based Mobility Model [5] is employed in the analysis. The problem setup consists of an ad hoc network with a number of nodes moving independently on a 2-dimensional torus in a geographical region, and each node belongs to a predetermined community. Each node can transmit up to a distance $K \geq 0$ meters away, and each message forwarding (in one-hop) takes one time unit.

Euclidean distance is used to measure the proximity between two nodes (or their positions) A and B. A slotted collision avoidance MAC protocol with Clear-to-Send (CTS) and Request-to-Send (RTS), is implemented for contention resolution. A message is acknowledged if it is received successfully at the encountered node by sending back a small acknowledgment packet to the sender.

The performance measures in the analysis include the average delivery probability and the message delivery delay.

A. Delivery Probability

In order to calculate the expected message delivery ratio, any path of message *m* between S and D is a *k - hop* simple path, denoted as *l*, which is represented by a set of nodes and links denoted as $\{S; h_1 h_2 \ldots, h_{k-1}; D\}$, and $\{e_1, e_2, \ldots, e_k\}$, respectively. The cost on each edge, denoted as $\{\beta_1, \beta_2, \beta_3, \ldots, \beta_k\}$, is the inter-contact rate (or frequency) of each adjacent node pair along the path. According to the forwarding policy of SAURP, the values of inter-contact rate should satisfy $\{\beta_1 < \beta_2 < \beta_3 < \ldots < \beta_k\}$. The path cost, $PR_l(t)$, is the probability that a message *m* is successfully forwarded from S to D along path *l* within time *t*, which represents a cumulative distribution function (CDF). The probability density function of a path *l* with *k - hop* for one message copy can be calculated as convolution of k probability distributions [15] which is calculated as:

$$P_{rl}(t) = p_1(t) \otimes p_2(t) \otimes \ldots p_k(t) \qquad (9)$$

The probability of message delivery on forwarding path *l* between any source S, and destination D, within expiration time T is expressed as:

$$F_l(T) = PR_l(Td_l < T) = \int_0^T P_{rl}(t)dt$$
$$= \sum_{i=1}^{k1} C^{kl}_i \int_0^T Pi(t)dt$$

$$PR_l(Td_l < T) = \sum_{i=1}^{K1} C^{(kl)}_i :(1 - e^{-\beta i T}) \qquad (12)$$

If there are L-1 copies (excluding the message at the source) of message m traversing through L-1 independent paths in the network, the maximum probability of message delivery can be written as

$$PR_{max}(T_d < T) = \max\{PR_{SD}, PR_1, PR_2, \ldots, PR_{L-1}\} \qquad (13)$$

where PR_{SD} and PR_l are random variables representing the delivery probability in case of direct message delivery between S and D, and through one of L - 1 paths, respectively. The expected delivery probability of message m with L - 1 copies traversing on L - 1 paths is calculated as:

$$PR(Td < T) = 1 - PRSD (TSD > T) \prod_{l=1}^{L-1}(1 - PR_l(Td_l < T)(14)$$

$$PR(T_d < T) = 1 - e^{-\beta SDT} \prod_{l=1}^{L-1} \left(\sum_{i=1}^{k1} C^{(k1)}_i (e^{-\beta i T}) \right) \qquad (15)$$

By assuming X totally generated messages in the network, the average of the delivery probability in the network is calculated as

$$PR = 1/X \sum_{m=1}^{X} PR_m \qquad (16)$$

B. Delivery Delay

Let message *m* have L - 1 copies (excluding the message at the source) traversing on L - 1 independent paths. The minimum delivery delay can be written as:

$$D_{SD} = min\{T_{SD}, Td_1, Td_2, \ldots Td_{L-1}\} \qquad (18)$$

where T_{SD} and Td_l are a random variables representing the delivery delay through direct path between S and D and through one of $L\square1$ paths, respectively. The expected delay of message m, $E[D_{SD}]$, can be calculated as

$$E[D_{SD}] = \int_0^\infty P(T_d > t) = \int_0^\infty e^{-\beta SDt} \prod_{l=1}^{L-1}(\sum_{i=1}^{k1} C^{(k1)}{}_i \cdot e^{-\beta it}) \, dt$$
$$= 1/\beta_{SD}\int_0^\infty \beta_{SD} e^{-\beta SDt}\prod_{i=1}^{L-1}(\sum_{i=1}^{K1} C^{(k1)}{}_I \cdot e^{-\beta it}) \, dt =$$
$$1/\beta_{SD} E\{\prod_{i=1}^{L-1}(\sum_{i=1}^{k1} C^{(k1)}{}_I \cdot e^{-\beta iTSD})\} \, , \, T_{SD} < \infty \quad (19)$$

The above relation gives an upper bound on the delivery delay since it is conditioned to T_{SD}, $T_{SD} < \infty$ and can be taken as point of reference.

The average delivery delay of message *m* can be calculated intuitively as:

$$E[ED_{(S,D)}] = [\, 1/L \, (T_{SD} + \sum_{i=1}^{L-1} Td_l \,)]. \quad 1/ \, PR(T_d < T) \quad (20)$$

T_{SD} is included in (19) only if $T_{SD} < \infty$.
By assuming X totally generated messages in the network, the average delivery delay can thus be calculated as

$$DR = 1/X \sum_{m=1}^{x} D_m \qquad (21)$$

C. Validation of Analytical Model

In order to evaluate the accuracy of the mathematical expressions in this analysis, SAURP is examined under two network status scenarios. In the first scenario, the network is operating under no congestion, i.e., all the nodes have infinite buffer space, and the bandwidth is much larger than the amount of data to be exchanged between any two encountered nodes. In the second scenario, the network is operating under limited resources, i.e., the forwarding opportunities can be lost due to high traffic, limited bandwidth, limited buffer space, or contention (i.e., more than one node within the transmission range are trying to access the wireless channel at the same time). For both scenarios, 50 nodes move according to community-based mobility model [5] in a 300x300 network area. The transmission range is set to 30 to enable moderate network connectivity with respect to the considered network size. The traffic load is varied from a low traffic load (i.e., 20 messages generated per node in 40,000 time units) to high traffic load (i.e., 80 messages generated per node in 40,000 time units). A source node randomly chosen a destination and generates messages to it during the simulation time. In this analysis the message copies are set to 5 (i.e., forming a maximum of 5 paths). To enable accurate analysis, the simulation program is run for a period of time (warm up period of 10,000 time units) such that each node can build and maintain the best forwarding paths with every other node in the network.

In this analysis, we simplified the calculation by limiting our study to only the best two of forwarding paths among all other paths and compare the simulation and theoretical results of delivery ratio and delivery delay. In most cases, a message takes the best forwarding path that based on the inter-encounters history if the network is not congested and the buffers operate under their capacity limit.

When the network resources are enough to handle all the traffic loads (Scenario 1), there is no dramatic change in the obtained delivery ratio and delivery delay for all traffic loads. That is because messages follow the best forwarding paths that lead to best performance. The simulation and analytical plots for SAURP present close

match and validates the generality of the analytical expressions. Additionally, it is evident that (16) and (20) are tight for all degrees of traffic loads. When the network resources are limited (i.e, scenario 2), the contention and the overhead of MAC layer increase, resulting in longer forwarding paths, higher drop rate, and longer delivery delay. The simulation and analytical plots are still providing close match with small diverge in case of high traffic loads.

Although the contention does affect the accuracy of our theoretical expressions, the error introduced for SAURP is not large (20%), even for large traffic loads. Therefore, we believe the analytical expression is useful in assessing the performance in more realistic scenarios with contention.

V. PERFORMANCE EVALUATION

A. Experimental Setup

To evaluate the SAURP, a DTN simulator similar to that in [16] is implemented. The simulations are based on two mobility scenarios; a synthetic one based on community based mobility model (CBMM) [5], and a real-world encounter traces collected as part of the Infocome 2006 experiment, described in [20].

The problem setup consists of an ad hoc network with a number of nodes moving independently in a geographical region, and each node belongs to a predetermined community. Each node can transmit up to a distance $K \geq 0$ meters away, and each message transmission takes one time unit. A slotted collision avoidance MAC protocol with Clear-to-Send (CTS) and Request-to-Send (RTS), is implemented for contention resolution. A message is acknowledged if it is received successfully at the encountered node by sending back a small acknowledgment packet to the sender. The performance of SAURP is examined under different network scenarios and is compared with some previously reported schemes listed below.

- PROPHET [3]
- Spray and Focus (S&F) [5]
- Most mobile first (MMF)[14]
- Delegation forwarding (DF) [9]
- Self-Adaptive utility-based routing protocol (SAURP)
- Self-Adaptive routing protocol (SARP) [10]

The performance comparison under the considered mobility scenarios is in terms of average delivery delay, delivery ratio, and the total number of transmissions performed for all delivered messages.

B. CBMM Scenario

1) Evaluation Scenarios
In the simulation, 110 nodes move according to the community-based mobility model [5] in a 600 x 600 meter network in a given geographical region. The simulation duration is 40,000 time. The message inter-arrival time is uniformly distributed in such a way that the traffic can be varied from low (10 messages per node in 40,000 time units) to high (70 messages per node in 40,000 time units). The message time to live (TTL) is set to 9,000 time units. Each source node selects a random destination node, begins generating messages to it during simulation time.

We analyze the performance implication of the following. First, the performance of the protocols is evaluated with respect to the impact of the number of

message copies. Second, with respect to the low transmission range and varying buffer capacity under high traffic load. Third, with respect to the moderate-level of connectivity and varying traffic load. Fourth, the performance of the protocols is examined in terms of the bandwidth. Finally, the performance of the protocols is examined in terms of the level of connectivity changes.

Impact due to Number of Message Copies

The transmission range K of each node is set to 30 meters, leading to a relatively sparse network. In order to reduce the effect of contention on any shared channel, the traffic load and buffer capacity is set to medium (i.e., 40 generated messages per node) and high (i.e. 1,000 messages per node), respectively. The number of message copies is then increased from 1 to 20 in order to examine their impact on the effectiveness of each protocol. The proposed SAURP is compared with the S&F and MMF schemes, since each scheme has a predefined L to achieve the best data delivery. Note that the value of L depends on the application requirements, the mobility model considered, and the design of the protocol.

The Effect of Buffer Size

In this scenario the performance of SAURP regarding different buffer sizes is examined under a low transmission range (i.e., K = 30) and a high traffic load (i.e., 50 messages generated per node). Due to the high traffic volumes, we expect to see a significant impact upon the message forwarding decisions due to the degradation of utility function values caused by buffer overflow. Note that when the buffer of the encountered node is full, some messages cannot be delivered even though the encountered node metric is better than the custodian node. This situation results in extra queuing delay, especially in the case that flooding-based schemes are in place.

It is observed that the SAURP scheme produced the best performance in all scenarios, since it takes the situation that a node may have a full buffer into consideration by degrading the corresponding utility metric, it produced the best performance.

The Effect of Traffic Load

The main goal of this scenario is to observe the performance impact and how SAURP reacts under different degrees of wireless channel contention. The network connectivity is kept high (i.e., the transmission range is set to as high as 70 meters) under different traffic loads, while channel bandwidth is set relatively quite small (i.e., one message transfer per unit of time) in order to create congested environment. We have two scenarios for nodal buffer capacity: 1) unlimited capacity; and 2) low capacity (15 messages).

It is observed that PROPHET produced the largest delivery delay and requires a higher number of transmissions compared to all the other schemes. PROPHET produced an order of magnitude more transmissions than that by SAURP.

When the traffic load is increased, the available bandwidth will decreased accordingly, which causes performance reduction. When the traffic load is moderate (i.e., less that 50 messages), it was clear that the delivery delay is short in all the schemes, while SAURP outperforms

all other protocols and MMF is the second best. This is because in MMF, the effect of buffer size is relaxed, which makes nodes buffer an unlimited number of messages while roaming among communities. SAURP can produce delay shorter than that of PROPHET, MMF, DF, S&F, and SARP by 350%, 52%, 400%, 250%, and 57%, respectively. Regarding the delivery ratio, SAURP, MMF, S&F, and SARP can achieve excellent performance of 98%, while the PROPHET routing degrades below 60% for high traffic loads. DF can achieve delivery ratio above 92%.

We observed that SAURP can achieve significantly better performance compared to all the other schemes, due to the consideration of busy links in its message forwarding mechanism, where the corresponding routing-metric is reduced accordingly. This results in the ability of rerouting the contended messages through the areas of low congestion.

As the buffer capacity is low (e.g., 15 messages) and the traffic load is high, the available bandwidth decreases and the buffer occupancy increases accordingly, which makes the performance of all protocols degraded, especially for the PROPHET and MMF.

The Effect of Channel Bandwidth and Traffic Load

To examine the effect of channel bandwidth, the network connectivity is set to moderate (under moderate transmission range by setting K = 50), and the link capacity is set five times higher than that used in the previous scenarios in order to avoid bottlenecks in the traffic loads.

As the link bandwidth increased the performance of all routing schemes will improved with respect to delivery delay and delivery ratio, because the buffer capacity is unlimited and the contention on the bandwidth is relaxed. SARP achieve the best performance, while SAURP achieves the second best compared to the other schemes. It outperforms MMF scheme, since MMF is coupled by the number of message copies. Compared to PROPHET, DF, and S&F, SAURP has a shorter delay by 450%, 390%, and 83%, respectively. Meanwhile, SAURP needed much less transmissions compared to that by S&F. Even though DF produced the lowest number of transmissions, it has the worst performance in terms of delivery delay and delivery ratio. All protocols achieved a delivery ratio above 90%. Compared to other protocols, SAURP maintains the second highest delivery ratio after SARP: above 96%.

The Effect of Connectivity

This scenario studies the performance impact due to network topology connectivity. In the scenario, the level of connectivity is increased from very sparse to highly connected by varying the value of K while observing the resultant impact on the performance. We are particularly interested to investigate the SAURP mechanism in response to heavy traffic loads which result in high contention on the wireless channel. The buffer capacity is kept low (15 messages), and the traffic load is considerably high (60 messages per node).

SAURP outperforms all the schemes in terms of delivery delay while taking noticeably fewer transmissions than that by S & F and and SARP schemes under all connectivity considered in the simulation. When the network is sparsely connected, SAURP can achieve shorter delivery delay than all other schemes, that is because the performance of other schemes is affected by the uncertainty

of buffer occupancy status. On the other hand, when the network is moderate-connected, SARP can achieve commutative-level of delivery delay compared to SAURP with more transmissions. As the network becomes almost connected and the traffic load is high, the uncertainty of both buffer occupancy status and the availability of bandwidth affect the performance of the other techniques. As a result, SAURP outperforms all other schemes in terms of delivery delay and delivery ratio.

C. Real Trace Scenario

In order to evaluate SAURP in realistic environment, the performance of the scheme is examined using real encounter traces. These data sets comprise of contact traces between short-range Bluetooth enabled devices carried by individuals in Infocome 2006 conference environment. In order to observe the performance impact and how SAURP reacts under congested environment, we set the bandwidth, buffer capacity, and the distribution of the contact time such that congested environment is formed. The channel bandwidth is set relatively quite small (i.e., one message transfer per unit of time), and the buffer size is set to 10, under different levels of traffic demand.

As the buffer capacity is low and the traffic load is high, the available bandwidth decreases and the buffer occupancy increases accordingly, which makes the performance of all protocols degraded, especially for the PROPHET and MMF. It is observed that PROPHET produced the largest delivery delay. It is subject to at least 2.1 times of longer delivery delay than that by SARP. It is notable that SAURP outperforms all the multiplecopy routing protocols in terms of delivery delay and delivery ratio under all possible traffic loads. When the traffic load is high, SAURP yielded shorter delivery delay than that of MMF by 52%, SF by 30%, and DF by 40%. Although SAURP requires more transmissions compared to the MMF and DF, the number is still smaller than that produced by S&F. SAURP can achieve delivery ratio above 76% for high traffic loads, while the SARP, POPHET, DF, S&F, and MMF degrades by 67%, 38%, 53%, 60%, and 50%, respectively.

VI. CONCLUSION

The paper introduced a novel multi-copy routing scheme, called SAURP, for intermittently connected mobile networks that are possibly formed by densely distributed and hand-held devices such as smart phones and personal digital assistants. SAURP aims to explore the possibility of taking mobile nodes as message carriers in order for end-to-end delivery of the messages. The best carrier for a message is determined by the prediction result using a novel contact model, where the network status, including wireless link condition and nodal buffer availability, are jointly considered. We provided an analytical model for SAURP, whose correctness was further verified via simulation. We further compared SAURP with a number of counterparts via extensive simulations. It was shown that SAURP can achieve shorter delivery delays than all the existing spraying and flooding based schemes when the network experiences considerable contention on wireless links and/or buffer space. The study provides significance that when nodal contact does not solely serve as the major performance factor, the DTN routing performance can be significantly improved by further considering other resource limitations in the utility function and message weighting/forwarding process.

REFERENCES

[1] Disruption Tolerant Networking. http://www.dtnrg.org.
[2] A. Lindgren, A. Doria, and O. Schelen. Probabilistic routing in intermittently connected networks. SIGMOBILE Mobile Comput. Commun. Rev., 7(3):19–20, 2003.
[3] A. Vahdat and D. Becker, "Epidemic Routing for Partially Connected Ad Hoc Networks ," Duke University, Tech. Rep. CS-2000- 06, July 2000.
[4] T. Spyropoulos, K. Psounis, and C. S. Raghavendra. "Efficient Routing in Intermittently Connected Mobile Networks: The Single-copy Case," IEEE Trans. Networking Vol. 16, Issue. 1, Feb. 2008
[5] _____, "Efficient routing in intermittently connected mobile networks: The Multi-copy case," IEEE Trans. Networking vol. 16, Issue 1, Feb. 2008.
[6] Z. Li and Haiying Shen. Utility-based Distributed Routing in Intermittently Connected Networks. 37th International Conference on Parallel Processing. (ICPP '08).September 8-12, 2008, Portland, Oregon, USA.
[7] T. Spyropoulos, K. Psounis, and C. S. Raghavendra. "Utility-based Message Replication for Intermittently Connected Heterogeneous Wireless Networks", in Proceedings of IEEE WoWMoM workshop on Autonomic and Opportunistic Communications (AOC), (INRIA Technical Report RR-6129), June 2007.
[8] J. Broch, D. A. Maltz, D. B. Johnson, Y.-C. Hu, and J. Jetcheva. A performance comparison of multi-hop wireless ad hoc network routing protocols. Proc. ACM MobiCom. Oct. 1998
[9] V. Erramilli, M. Crovella, A. Chaintreau , C. Diot. Delegation forwarding. Proceedings of the 9th ACM international symposium on Mobile ad hoc networking and computing, May 26-30, 2008, Hong Kong, Hong Kong, China.
[10] Ahmed Elwhishi and Pin-Han Ho. 2009. SARP - a novel multi-copy routing protocol for intermittently connected mobile networks. In Proceedings of the 28th IEEE conference on Global telecommunications (GLOBECOM'09), Mehmet Ulema (Ed.). IEEE Press, Piscataway, NJ, USA, 4482-4488.
[11] Musolesi, M., and Mascolo, C. 2009. CAR: Context-aware Adaptive Routing for Delay Tolerant Mobile Networks. IEEE Transactions on Mobile Computing. 8, 2 (February 2009), 246-260.
[12] Q. Yuan, I. Cardei, and J. Wu. 2009. Predict and relay: an efficient routing in disruption-tolerant networks. In Proceedings of the tenth ACM international symposium on Mobile ad hoc networking and computing (MobiHoc '09).
[13] A. Balasubramanian , B. Levine , A. Venkataramani, DTN routing as a resource allocation problem, Proceedings of the 2007 conference on Applications, technologies, architectures, and protocols for computer communications, August 27-31, 2007, Kyoto, Japan.
[14] T. Spyropoulos, T. Turletti, and K. Obraczka, "Routing in delaytolerant networks comprising heterogeneous node populations," IEEE Transactions on Mobile Computing, vol. 8, no. 8, pp. 1132–1147, August 2009.
[15] Akkouchi M. On the convolution of exponential distributions. J. Chungcheong Math. Soc, vol. 21 no.4. pp.501–510, 2008.
[16] "DTN java simulator," http://people.ee.ethz.ch/ spyropot/ dtnsim.html.
[31] Daly, E.M.; Haahr, M.; , "Social Network Analysis for Information Flow in Disconnected Delay-Tolerant MANETs," Mobile Computing, IEEE Transactions on , vol.8, no.5, pp.606-621, May 2009 doi: 10.1109/TMC.2008.161.
[17]] K. Tan, Q. Zhang, andW. Zhu, "Shortest path routing in partially connected ad hoc networks," in Globecom, 2003.
[18] MaxProp: J. Burgess, B. Gallagher, D. Jensen, and B. N. Levine, "MaxProp: Routing for vehicle-based disruption-tolerant networks," in Proc. of INFOCOM, 2006.
[19] MOPS: F. Li and J. Wu, "Mops: Providing content-based service in disruptiontolerant networks," in Proc. of ICDCS, 2009, pp. 526–533.SimBet: E. M. Daly and M. Haahr, "Social network analysis for routing in disconnected delay-tolerant manets," in Proc. of MobiHoc, 2007.
[20] CHAINTREAU, A., HUI, P., CROWCROFT, J., DIOT, C., GASS, R., AND SCOTT, J. Impact of Human Mobility on Opportunistic Forwarding Algorithms. IEEE Trans. on Mobile Computing 6, 6 (2007), 606–620.