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Scalable Reliable Multicasting in Delay Tolerant Networks with Erasure Correction

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Abstract *:* Delay Tolerant Networks (DTNs), also called as intermittently connected mobile networks, are wireless networks in which a fully connected path from source to destination is unlikely to exist. However, effective forwarding based on a limited knowledge of contact behavior of nodes is challenging. When large files need to be transferred from source to destination make all the packets available at the source and transfer the file as small packets. We study the packets arrival at source and analysis their performance. We consider the linear blocks and rate less linear coding to generate redundancy and also for energy constraint .We scheduling the large file into small packets and delivering through multipath to destination, for this we use optimal user centric allocation and scheduling the packets in the receiver side.

Key words: Delay tolerant networks, network coding, rate less codes,.

INTRODUCTION

Delay Tolerant Networks (DTNs), also called as intermittently connected mobile networks, are wireless networks in which a fully connected path from source to destination is unlikely to exist. In these networks, for message delivery , nodes use store-carry-and-forward paradigm to route the messages. The examples of this networks are wildlife tracking, military networks etc. However, effective forwarding based on a limited knowledge of contact behavior of nodes is challenging. it becomes crucial to design efficient resource allocation and data storage protocols. Although the connectivity of nodes is not constantly maintained, it is still desirable to allow communication between nodes. Each time the source meats a relay node, it chooses a frame for transmission with probability In the basic scenario, the source has initially all the packets .

Under this assumption it was shown in that the transmission policy has a threshold structure: it is optimal to use all opportunities to spread packets till some time σ depending on the energy constraint, and then stop. This policy resembles the well-known "Spray-and-Wait" policy. In this work we assume a more general arrival process of packets: they need to be simultaneously available for transmission initially, i.e., when forwarding starts, as assumed in . This is the case when large multimedia files are recorded at the source node that sends them out (in a DTN fashion) after waiting for the whole file reception. This paper focus es on general packet arrivals at the source and two-hop routing. We distinguish two cases: when the source can overwrite its own packets in the relay nodes, and when it cannot.

DTNs exploit random contacts between mobile nodes to allow end-to-end communication between points that do not have end-to-end connectivity at any given instant. This is obtained at the cost of replications of data and hence of energy and memory resources. To transfer successfully a file, all frames of which it is composed are needed at the destination. The memory of a DTN node is assumed to be limited to the size of a single frame. We study adding coding in order to improve the storage efficiency. We consider Reed-Solomon type codes as well as network coding. The basic questions are Then: (i) transmission policy: When the source is in contact with a relay node, should it transmit a frame to the relay? (ii) Scheduling: If yes, which frame should a source transfer? Each time the source meats a relay node, it chooses a frame i for transmission with probability u I In a simple scenario, the source has initially all the frame and u I are fixed in time. It was shown in that the transmission policy has a threshold structure: use all opportunities to spread frame till some time σ and then stop (this is similar to the "spray and wait" policy).

Due to convexity arguments it turns out that the optimal u I does not depend on i In this paper we assume a general arrival process of frames: they need not become available for transmission simultaneously at time zero as in. We further consider dynamic scheduling: the probabilities u I may change in time. We define various performance measures and solve various related optimization problems. Surprisingly, the transmission does not follow anymore a threshold policy (in contrast with). We extend these results to include also coding, and show that all performance measures improve when increasing the amount of redundancy. We then study the optimal transmission under network coding.

Related work

In forward error correction (FEC)method Several satellites need to receive several data packets, that may need retransmission due to channel errors Because many sites may need retransmissions, the problem is to avoid the phenomenon of ACK implosion due to several sites requesting repairs several works to combine FEC and acknowledgment-based retransmission protocols, such as The effort there was to improve timeliness of packet delivery in multicasting multimedia streams which are subject to hard delay constraints. In DTNs the framework is different since the challenges are to overcome frequent disconnections. Papers and propose a technique to erasure code a file and distribute the generated code -blocks over a large number of relays in DTNs, so as to increase the efficiency of DTNs under uncertain mobility patterns. In the performance gain of the coding scheme is compared with simple replication. The benefit of coding is assessed by extensive simulations and for different routing protocols, including two hop routing.

The works and describe the technique to erasure code a file and distribute the generated code-blocks over a large number of relays in DTNs. The use of erasure codes is meant to increase the efficiency of DTNs under uncertain mobility patterns. In the performance gain of the coding scheme is compared to simple replication, i.e., when additional copies of the same file are released.

The benefit of erasure coding is quantified by means of extensive simulations and for different routing protocols, including two-hops routing. In, the authors address the case of non-uniform encounter patterns, and they demonstrate strong dependence of the optimal successful delivery probability on the way replicas are distributed over different paths. The authors evaluate several allocation techniques; also, the problem is proved to be NP–hard. The paper proposes general network coding techniques for DTNs. In ODE based models are employed under epidemic routing; in that work, semi-analytical numerical results are reported describing the effect of finite buffers and contact times; the authors also propose a prioritization algorithm. The paper addresses the design of stateless routing protocols based on network coding, under intermittent end-to-end connectivity. A forwarding algorithm based on network coding is specified, and the advantage over plain probabilistic routing is proved when delivering multiple frames.

Proposed System

Consider a network that contains N + 1 mobile node. Two nodes are able to communicate when they come within reciprocal radio range and communications are bidirectional .We assume that the duration of such contacts is sufficient to exchange all frames: this let us consider nodes meeting times only, i.e., time instants when a pair of not connected nodes fall within reciprocal radio range. Time between contacts of pairs of nodes is exponentially distributed with given inter-meeting intensity. A file contains K frames. The source of the file receives the frames at some times are called the arrival times.

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The transmitted file is relevant during some time τ . By that we mean that all frames should arrive at the destination by time t 1+τ. We do not assume any feedback that allows the source or other mobiles to know whether the file has made it successfully to the destination within time τ. If at time t the source encounters a mobile which does not have any frame, it gives it frame i with probability ui(t). Consider two-hop routing In this we used two concept overwrite case and non-overwriting case .In the existing concept non-overwriting case are highly efficient but overwriting case without constraints are not efficient, so in this work we use rate less code and block code for removing the overwriting case due to the transmission of packet .

Rate less code and block code is used for share the information sequence to the receiver without data loss, overwriting and delay. In this work due to the data transmission the multi path can be create using optimal user centric algorithm in the source side. Using the multi path the data can split into packet and assign packet to each node due to the transmission then packet are schedule using decentralized routing process based on the integer linear programming in the receiver side In the scheduling packet the packet can schedule and receive to the client side. We use erasure coding technique to increase the reliability and to further decrease the cost of routing.

For a given desired delivery rate and deadline for delivery, we find the optimum parameters to obtain the smallest cost both in single period and two period erasure coding based routing. We also analyze the effects of message distribution algorithms on the cost of routing both in replication based (i.e. spray and wait) and erasure coding based algorithms. We analyze real DTN traces and detect the correlations between the movements of different nodes using a new metric called conditional intermeeting time. We then use the correlations between the meetings of a node with other nodes for making the existing single -copy based routing algorithms more costeficent.algorithms

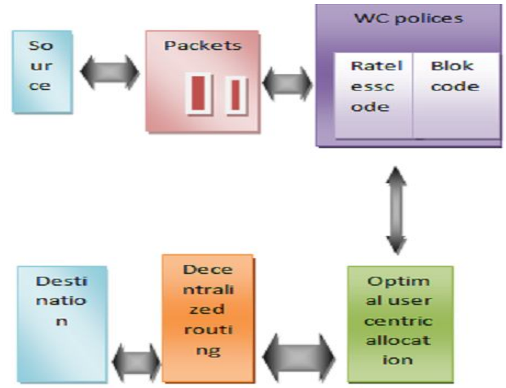

Fig1.Packets arrivals

Scalable Reliable Multicast

There have been many reliable multicast schemes proposed, but most of them cannot scale to large receiver sets. The reason they cannot scale usually boils down to problems relating to state explosion or message implosion.

For example, a sender-initiated scheme will require the sender to keep state information for each receiver. As the receiver set grows, this state can become too large to store or manage – a state explosion. Message implosion occurs when receivers send packets back to the sender. With a large number of receivers there is a danger that this traffic will overwhelm the sender, or the network links leading to the sender. For instance, a scheme that requires every receiver to send an acknowledgement for each packet received clearly will not scale to large receiver sets without implosions.

There are four key approaches to making reliable multicast scalable:

1. Error/Erasure correction
2. Hierarchy/localization
3. Suppression
4. Polling

SRM

SRM uses suppression of NACKs and of re-sends to avoid implosion. The original delay before sending a NACK is set as a random value in the range [c1d, c2d], where c1 and c2 are constants and d is an estimate of the one-way delay from the source to the receiver. When a NACK is sent, the timer is doubled to wait for the reply. If a host receives a NACK from another host for the missing data before its timer for sending a NACK goes off, it will do an exponential random back-off to wait for the data. That is, the i'th time it sets its timer for a particular packet, it will set the timer in the range 2i-1[c1d,

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c2d]. Any host that has a copy of the data can answer the NACK by re-transmitting the packet. When a host B receives a NACK from host A that has the data to respond to, it sets a repair timer in the range [b1dAB, b2 dAB] where b1 and b2 are constants and dAB is the estimated one-way delay between A and B. If host B receives a repair for the missing data before its timer goes off, it will cancel the repair, otherwise it will send the data when the timer goes off.

In order to prevent duplicate NACKs from triggering duplicate replies, host B will ignore NACKs for the same packet for 3 dSB , where dSB is the one-way delay from the original source to B. SRM suffers in two respects: scalability, and rate control. SRM requires one-way delay estimates between all nodes in a session. These are calculated as ½ round trip time. With n nodes, this requires either O(n2) messages or O(n) messages of size n. In either case, O(n2) traffic is required to calculate round-trip time. Clearly, this cannot scale to accommodate arbitrarily large receiver sets.

SRM's random delays are based purely on inter-node delay estimates, not on the audience size. Therefore, if the audience size is arbitrarily increased without a corresponding delay increase, implosion will eventually result. It is possible to introduce more delay to increase scalability, but adding delay is only useful to a point; eventually delays become longer than the session itself

SRM lacks of control over the rate of incoming data to a given receiver, due to its distributed nature. At a given time several nodes may be sending NACKs, several may be re-sending data, and the sender may be sending new data. Even with the sender controlling the rate at which it sends new data, the re-sends and NACKs may lead to an aggregate traffic level, which is higher than some nodes can handle. This is of particular concern when some receivers have a modem connection, in which case their bandwidth limitation is very strict, and relatively low.

Erasure Correcting SRM

Erasure Correction SRM (ECSRM) modifies SRM to address some of the problems noted above. To deal with the lack of rate control, ECSRM only allows the sender to perform re-sends. Rate control is applied as part of ECSRM to ensure that the aggregate of new sends and re-sends does not exceed the desired rate. NACK messages will still add to the rate, but due to their small size and NACK suppression, their impact should be limited. Having only the sender perform re-sends also means there is no need for calculating round trip time, which eliminates the O(n2) traffic required for this calculation. This also eliminates a feature of SRM, namely that a node close to the point of loss is most likely to perform the re-send.

However, we would point out that:

• This feature is only probabilistic, not guaranteed

• The node nearest the point of loss may not really be suitable – it may be under-powered or on a slow/congested link

• We assume that applications using ECSRM can afford to trade off time for scalability – so fast re-sends are not critical

• In the case of live telepresentations, continued connectivity to the sender is critical.

Additional fault tolerance from this feature is not important.

CONCLUSIONS

We have addressed the problem of optimal transmission and scheduling policies in DTN with two-hop routing under memory and energy constraints, when the packets of the file to be transmitted get available at the source progressively. We solved this problem when the source can or cannot overwrite its own packets, and for WC and non WC policies memory and energy constraints, when the packets of the file to be transmitted get available at the source progressively. We solved this problem when the source can or cannot overwrite its own packets, and for WC and non WC policies

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The preferred spelling of the word "acknowledgment" in American English is without an "e" after the "g." Use the singular heading even if you have many acknowledgments. Avoid expressions such as "One of us (S.B.A.) would like to thank ... ." Instead, write "F. A. Author thanks ... ." **Sponsor and financial support acknowledgments are placed in the unnumbered footnote on the first page**.

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