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Network Coding for TCP Flows

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Abstract— ***Network coding for wireless network has proved to be an effective tool in terms of throughput improvement. ACK-based flow control mechanisms forms a central part of today's IP such as TCP. The paper focuses on using NC with TCP for multi-hop networks for throughput improvement. Wireless networks support multi-gigabit flow rates along with the demands of running bandwidth intensive applications on mobile devices. Wireless channel quality changes dynamically due to multipath, doppler, interference, and other fading effects. The link bandwidth can vary, which causes variations on the bandwidth available to the mesh nodes and creates problems to the TCP-based packet flows. So, a new network layer approach, came into picture which consists ideas like interflow network coding, multiple-path routing, and distributed redundancy and buffer management to adaptively route packets from Internet to the corresponding destinations. This approach is fully distributed and can be integrated and deployed in existing protocol stack.***

Keywords— TCP/IP, Network Coding(NC), Bandwidth Delay Product (BDP), gigabit flows.

I. INTRODUCTION

Network coding[1] is a networking technique in which transmitted data is encoded and decoded to increase network throughput, reduce delays and make the network more robust. In network coding, algebraic algorithms are applied to the data to accumulate the various transmissions. The received transmissions are decoded at their destinations. This means sure that fewer transmissions are required to transmit all the data, but this requires more processing at intermediary and terminal nodes. There are many applications like VOD, HDTV, mobile internet access which requires more than 1Gbps bandwidth. Transmission Control Protocol (TCP) supports real-time streaming applications, which guarantees reliable and ordered packet delivery.

Wireless network is a promising approach to provide higher data rate and better performance in the future. In a wireless network, wireless hosts access the Internet infrastructure over a fixed multihop wireless network of "mesh nodes" . Each wireless host is associated with one of the mesh nodes and some mesh nodes called as gateway nodes have direct connection to Internet, as shown in Fig. 1. Therefore, downlink traffic from Internet to a particular wireless host enters the network through node G, traverses one or more hops in the mesh network to the corresponding access node (say node A) and then goes over the wireless link to the receiving host. Here,the idea is described in terms of the actions taken by the mesh nodes in forwarding TCP packets to the wireless hosts connected to the access node A. For clarity of presentation, the set of downlink TCP flows is referred as F_A. The channel quality in wireless networks today, fluctuates due to multipath, Doppler, interference, and other fading effects. As a result, there are variations on the bandwidth available to the mesh nodes,which creates distractions for the mesh nodes in wireless network. This in turn creates performance problems to the TCP based packet flows, that affects TCP throughput.

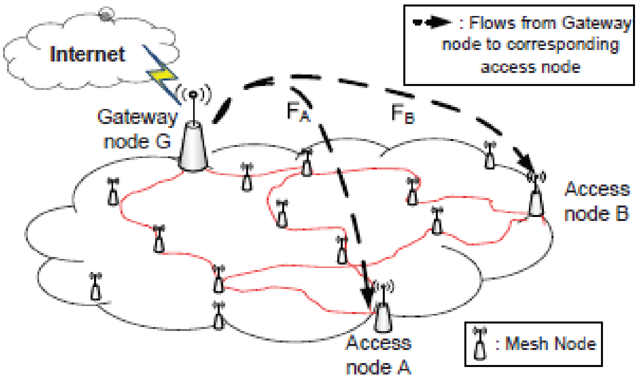

Fig 1: Wireless Network

This affects when BDP is large. BDP[5] is defined as the product of data link capacity in terms of bits per seconds (bps) and round trip delay time in seconds. Network Coding based approach is a network layer design to enhance the transport layer performance. It encodes packets from multiple flows. It adapts to the changes in available bandwidth. This method needs less re-routing as compared to previous ideas.

A. Motivation

The impact of the bandwidth fluctuations on TCP throughput is more significant when BDP is large. This is true not only for traditional TCP versions such as NewReno[7], but also for TCP versions specifically designed for large BDP networks such as Compound TCP (CTCP)[6].

There is a need for mechanisms supporting gigabit TCP throughput for each TCP flow in wireless networks. CTCP is not designed for handling the dynamic changes of link bandwidth. Improving TCP throughput requires additional strategies instantly and efficiently utilizing the available bandwidth.

B. Key idea

The key idea is in understanding that wireless networks are fundamentally different from wired networks. Wireless is a shared medium[16]. Wireless is stochastic in nature. Wireless is a broadcast medium. So, directly applying techniques from wired networks to wireless networks limits throughput and performance. Network coding promises a fundamentally new way to operate networks. At present, the intermediate nodes can only store, forward, or replicate the information they receive. This store-and-forward approach is closely related to the multi-commodity flow problem. Network coding questions the fundamental assumption in our store and forward network designs. The theory of network coding, was first introduced in seminal paper by Ahlswede et al. [1], breaks the convention of router networks, where more processing is required at intermediate and terminal nodes.

C. Goal: To develop strategies for improving end-to-end throughput.

Network Coding based approach[5] integrates the following ideas:

Digraph diversity (direct acyclic graph diversity) Inter-flow network coding with Spare bandwidth : Exploits excess bandwidth available on an outgoing link to introduce packet redundancy through network coding, to provide easy recovery from packet losses Buffer management: Ensures fair sharing of the limited storage available.

In this scheme, each gateway node forwards packets after network coding. Each intermediate node uses network coding before forwarding the packets to the outgoing links. Each mesh access node decodes the network coded packets before forwarding them to the destinations. The key feature of Network Coding based approach is that each node adapts its network coding rate based on the available bandwidth on the outgoing links, such that the access nodes can decode the packets. This approach also tries to reduce packet loss. This is achieved by sending the number of redundant packets to the networks based on the available bandwidth on its outgoing links.

II. RELATED WORK

The literature on network coding is extensive. Some papers deal with both network coding and TCP.

1. R. Ahlswede, N.Cai, S.-Y. R. Li, and R. W. Yeung, Network information flow, July. 2000.

Network coding[1] was first proposed for multicasting by Ahlswede in 2000, which aimed to increase data rate. Here, packets were encoded from he same flow.

2. X. Wang, T. T. Kwon, and Y. Choi, "A multipath routing and spectrum access (MRSA) framework for cognitive radio systems in multi-radio mesh networks, 2009.

The author X.Wang[2] proposed multipath routing. This approach is more prone to packet re-ordering. This happens when links used operate with unequal bandwidth along different routes. This re-routing was done based on information, collected from the nodes. This created delay which in turn reduced throughput.

3. J. K. Sundararajan, D. Shah, "Network coding meets TCP", Apr. 2009.

The author D. Shah[3] suggested a approach called- TCP/NC, a modification to protocol stack at the TCP sender and receiver. This approach introduced a new network coding sub layer between the transport and network layers, which was not desirable. Number of packets were sent according to the measured packet loss probability in the network. This was done to recover from packet loss.

4. K.-C. Leung, V. O. K. Li, and D. Yang, "An overview of packet reordering in Transmission Control Protocol (TCP): Problems, solutions, and challenges," Apr. 2007.

Packet reordering [4] was done when packets experiences different link delays through different routing paths. Packets were reordered before sending to the transport layer of the receiver. When transport layer receives consecutive packets, it increases its congestion window. But, packet losses causing packet transmission still reduced TCP throughput.

5. "Network Layer Support for Gigabit TCP Flows in Wireless Mesh Networks" Chin-Ya Huang, Parameswaran Ramanathan, IEEE, 2013

To alleviate the problem of low TCP throughput, we propose a Spare bandwidth Rate-adaptive Network Coding (SRNC) scheme. In SRNC scheme[5], each gateway node forwards packets after network coding. Each intermediate node also adaptively uses network coding before forwarding the packets to the outgoing links. Each mesh access node decodes the network coded packets before forwarding them to the destinations. The key feature of SRNC is that each node adapts its network coding rate based on the available bandwidth on the outgoing links, such that the access nodes can decode the packets with higher probability without significantly affecting the cross-traffic. This scheme shows that it uses the available bandwidth on each link efficiently and it significantly improves end-to-end throughput of TCP flows.

6. K. Tan, J. Song, Q. Zhang, and M. Sridharan, "A Compound TCP approach for high-speed and long distance networks", 2006

CTCP[6] is specifically designed for network with large BDP. It measures variance in RTT and uses this information to multiplicatively increase the sending rate with an objective of achieving high bandwidth utilization. CTCP is implemented in existing operating systems such as Windows 7, Windows Server 2008, and Linux. CTCP does not perform well as data rate increases.

7. S. Floyd and T. Henderson, "The NewReno modification to TCP fast recovery algorithm", 1999.

New Reno[7] is a traditional loss-based scheme. It detects one packet loss per RTT. It operates in fast recovery phase, until all the packets are acknowledged successfully. It makes use of ACK's as Partial ACK and Full ACK. It considers Partial ACK as a possibility of packet loss and stays in fast recovery phase, until Full ACK comes.

III. OVERVIEW Of NETWORK CODING

Network coding is a networking technique in which transmitted data is encoded and decoded to increase network throughput, reduce delays and make the network more robust. In network coding, algebraic algorithms are applied to the data to accumulate the various transmissions. The received transmissions are decoded at their destinations. This means that fewer transmissions are required to transmit all the data, as shown in Fig. 2, but this requires more processing at intermediary and terminal nodes.

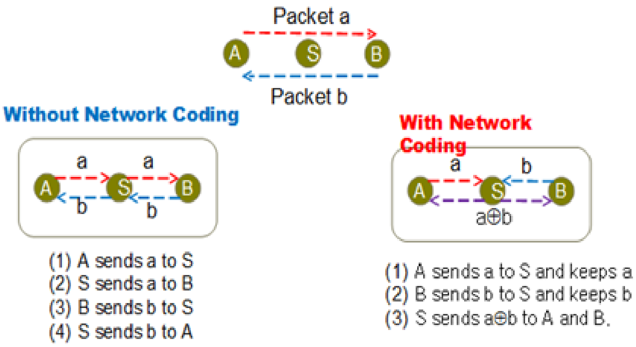

Fig. 2. Use of XOR to forward multiple packets

In traditional routing networks, packets are cached and forwarded downstream. Therefore, if a routing node receives two packets from two sources it forwards them one after another, and queues the others in the meantime, even if both are headed for the same destination. This requires separate transmissions for each and every message delivered, which decreases network efficiency. In network coding, algorithms are used to merge those two messages and the accumulated result is forwarded to the destination. After receiving the accumulated massage, it is decoded at the destination using the same algorithm. In order for this technique to work, the destination node needs to be completely synchronized with the transmitting nodes.

Network coding is perceived to be useful in:

1) Wireless mesh networks (WMN)
2) Messaging networks
3) Storage networks
4) Multicast streaming networks
5) File-sharing peer-to-peer networks

It also involves other networks where the same data needs to be transmitted to a number of destination nodes. The regular topology change that occurs in peer-to-peer networks poses a challenge to the network coding technique because it complicates network synchronization. In addition, the peers may need a large amount of processing time while trying to decode data. Overall, large networks can increase their efficiency through the use of network coding, but high overhead costs may make them less amenable for small networks.

Network coding is a technology of information switch which exploits coding and route. On the basis of conventional store-and-forward routing method, encoding received data packets can increase the information content per transmission and improve overall performance of the network. The essence is to use the nodes computing ability to improve bandwidth utilization of the link. Fig. 3.b. is the basic principle of network coding. In this figure, node S is the source, while X, Y denote the destination nodes, the bandwidth of each side is 1 bit per unit time, and now, two bits of data a and b will be sent from S to X and Y at the same time. It is easy to know that from S to X, Y, there are two independent paths, if using traditional routing method, as shown in Fig. 3. a. because there is a shared path WZ between the two paths ,a and b cannot be transmitted on link path WZ at the same time, so, the maximum information flow rate is 1.5 bits per unit time from S to X ,Y .

If using network coding method, on the node W, packets a and b can be performed XOR operations to be a new packet and then is delivered forward, the destination node X can recover the packet b by calculating (a XOR b XOR a) likewise destination. Coding is a technique that can be used to improve a network throughput, efficiency. In Network Coding, data is manipulated inside the network or at the network edges to achieve the maximum possible information flow in a network, based on principles of Shannon Information Theory. The above example uses a bitwise xor code, the coding operation could be much more general as:

The groups of bits as elements of a finite field, and a packet as a vector over this field.

Coding could then correspond to performing linear combinations of these vectors, with coefficients chosen from the field of operation.

In order to decode, the receiver will have to collect as many linear combinations as the number of packets that were mixed in, and then solve the resulting system of linear equations by Gauss Jordan elimination

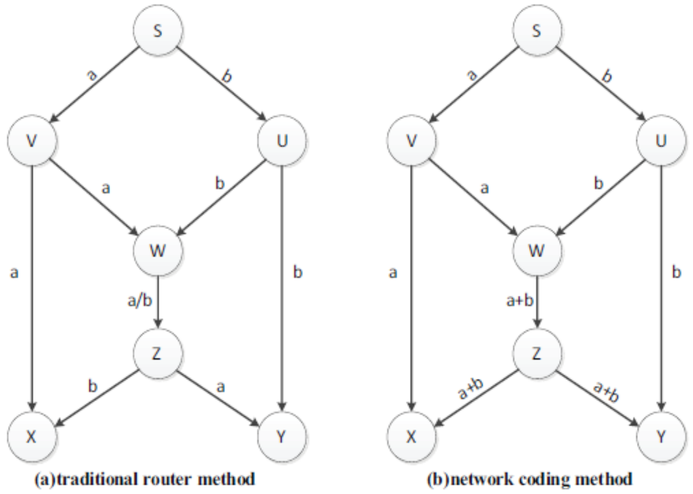

Fig. 3. Comparison between traditional routing and network coding method.

The throughput of today's wireless networks are far optimal. NC increases wireless throughput because it's broadcast nature coding allows the coders to compress the transmitted packets based on information that are known at various nodes. By matching what each neighbor has with what another neighbor wants, a coder can deliver multiple packets to different sources in a single transmission. This type of transformation is named inter-flow network coding because the coding is done over packets that differ in their nexthop, and thus from different flows.

IV. NETWORK CODING BASED APPROACH

Network Coding based approach is a network layer design required to enhance the transport layer performance. Transmission Control Protocol (TCP) is known to guarantee reliable and ordered packet delivery, commonly used for supporting real-time streaming. These applications also require low packet loss probability to sustain their Qualify of Service (QoS). Each node adapts its network coding rate based on the available bandwidth on the outgoing links[5]. In this scheme, each gateway node forwards packets after network coding. Each intermediate node also adaptively uses network coding before forwarding the packets to the outgoing links. Each mesh access node decodes the network coded packets before forwarding them to the destinations. It consists of following ideas, as shown in Fig. 4:-

Digraph acyclic diversity graph:

The routing algorithm is used. It finds a directed acyclic graph (DAG), at node Gateway node and ending at node Access node. Traditionally, packets of given flow are forwarded in the WMN over a single route. But, in the proposed approach, each gateway node forwards packets after network coding. This describes "Algo-rithm Gateway"[5], the steps taken by node G during the time interval [kT;(k + 1)T), where T is a pre-determined design parameter. Specifically, T is a design parameter chosen to ensure that the number of packets waiting for network coding is approximately equal to a desired target.

All packets transmitted during this interval [kT;(k + 1)T) has a special label with value k for other nodes to identify packets that are linear combination of the same set of packets using this label. During the interval [kT;(k + 1)T), G transmits linearly combined packets in FA arriving during the interval [(k − 1)T;kT). In order to assist buffer management policy, the first $n_{A;k}$ packets transmitted by G during this interval are marked as "High priority", and all the rest are marked as "Low priority".

Algorithm Gateway:

In time interval [kT; (k + 1)T)

1. Discard all packets in F_A which arrive in [(k -2)T; (k – 1)T).
2. Buffer all incoming packets in F_A.
3. At kT :
4. Let $n_{A,k}$ be the total number of packets in F_A which arrive in [(k-1)T; kT).
5. Create $n_{A,k}$ network encoded packets with linear combination in FA which arrive in [(k -1)T; kT).
6. Attach a label k to each of these packets.
7. Mark these network encoded packets as "High priority".
8. On each outgoing link, transmit (as permitted by the packet scheduler) network encoded packets :

9. If the total number of transmitted packets of all outgoing links is less than $n_{A,k}$,
10. Forward the “High priority” network encoded packets.
11. Else,

Generate a random linear combined packet of packets in FA. Attach a label k, mark it as “Low priority”, and forward it.

Inter-flow network coding with Spare bandwidth and Buffer management:

The intermediate node needs to computes a linear combination of packets in the given flow, i.e. it encodes packets. After encoding, intermediate node then forwards the resulting packets on one of the outgoing links in the DAG. Here, packets are network encoded across multiple flows (i.e. intersession network coding)[5]. This process relies on a self-clocking scheme with respect to labels. It also implements a buffer management strategy, where preference is given to “High priority” packets, if the buffer is full.

Algorithm at intermediate node i:

Part 1: When a packet in FA arrives on incoming link l:

// [steps 1 to 9] performs Buffer Optimization

1. If buffer is full;
2. If incoming packet is marked as “Low priority”,
3. Discard the incoming packet.
4. Else,
5. If all packets in buffer are marked as “High priority”,
6. Discard incoming packet.
7. Else,
8. Find the most recent “Low priority” packet in buffer and discard it.
9. Enqueue incoming packet.
10. Let L′ be the smallest unprocessed label of all FA packets in buffer.
11. Let Ll be the largest label of all F_A packets received on incoming link l.
12. When L′ <=μ min{Ll : incoming l}- 1:
13. k = L′.
14. Discard all packets in FA in buffer with label smaller than k.

Let n be the total number of packets in FA in buffer and n_h be the total number of “High priority” packets in buffer.

Part 2: On each outgoing link:

16. Transmit (as permitted by packet scheduler) linearly combined packets in F_A with label k in buffer no more than number of received packets of label k.
17. If the total number of packets sent by all outgoing links is less than n_h,
18. Duplicate an unsent “High priority” packet with label k and forward it.
19. Else,
20. Create a random linearly combined packet of packets in F_A in buffer Attach a label k, mark it as “Low priority”, and forward it.

Each node implements a buffer management strategy which ensures that no particular set of flows can dominate the buffer space at any node in the WMN. A full buffer causes packet losses to given flow, thereby decreasing the performance of flows. It hence ensures fair sharing of the limited storage.

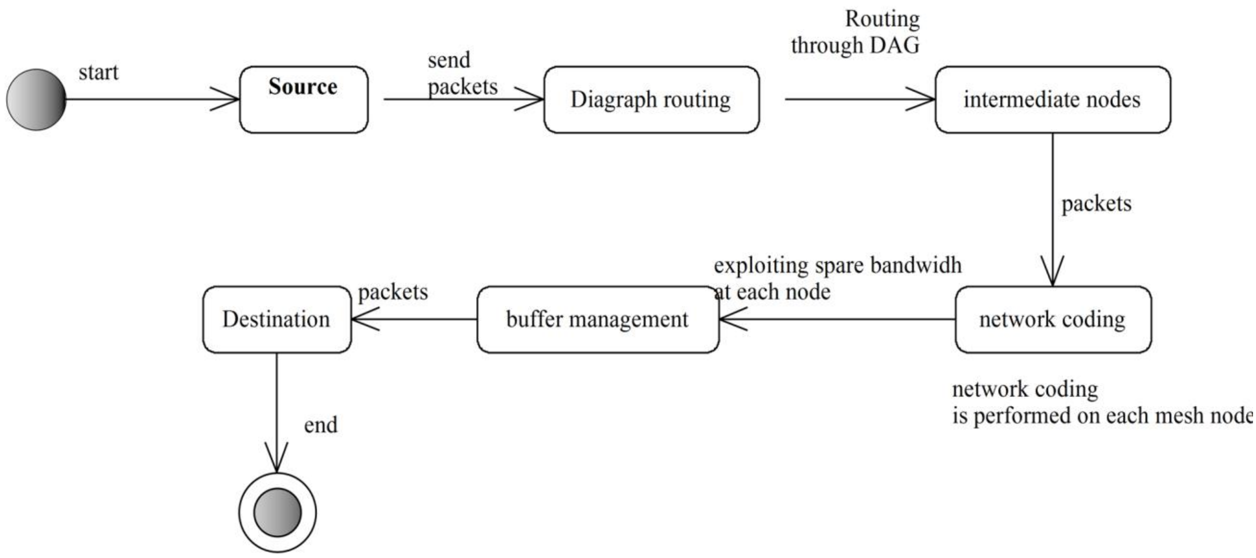

Fig 4: Network coding- based approach

Decoding at access node with Buffer management

Access node[5] implements a buffer management strategy similar to intermediate node. Access node also uses a self-clocking algorithm with respect to labels as intermediate node to determine which packets to decode. Here, access node which is associated to specific wireless host is responsible for decoding the network encoded packets.

Algorithm at access node i:

When a packet in FA arrives on incoming link l:

Steps [1 to 14] are same as that of Algorithm at intermediate node (I), then continue with the following steps as follows:

15. Apply decoding algorithm on packets in F_A with label k.
16. If the decoding algorithm succeeds,
17. Transmit the recovered unencoded packets on the outgoing links as per routing algorithm.
18. Else,
19. Discard undecoded F_A packets with label k.

If decoding is successful, the unencoded packets are forwarded to wireless hosts. Otherwise, undecoded packets are discarded. This scheme adapts the idea of network coding into unicast transmission. It network encodes packets from multiple flows. It adapts to the changes in available bandwidth. It needs much less re-routing along with previous ideas.

V. CONCLUSION

The network coding based scheme incorporates several ideas including interflow network coding, multiple path routing and distributed redundancy management to adaptively and reliably route packets from Internet to the corresponding destinations. Network Coding can be used for Cognitive radio enabled networks, data centers and priority based traffic management for real-time flows, etc. This idea of can be further extended to improve the end-to-end throughput in the underwater mesh networks.

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