Implementation of Optimal Pacing Scheme in Xinjiang's Oil and Gas Pipeline Leak Monitoring Network

Zhou Peng
College of Information Engineering, Tarim University, Alar Xinjiang, 843300, China
Email: zpzqxy@163.com

Abstract—Flow against Xinjiang's oil and gas pipeline leakage and the pipe network sudden burst pipe to pipeline leakage flow for the application objects, Optimal pacing scheme is designed in pipeline leak monitoring. Based on the property of Markov chain for network data, a new estimator with particle filter is proposed for congestion control in this paper. In the context of a reconfigurable transport protocol framework, we propose a QoS aware Transport Protocol (QSTP), specifically designed to operate over QoS (Quality of Service) enabled networks with bandwidth guarantee. The proposed scheme can adaptively adjust the network rate in real -time, so that it can efficiently avoid the traffic congestion. It proposes a Link Layer Adaptive Pacing (LLAP) scheme that adaptively controls the offered load into the network. The algorithms actively probe the underlay network and compute virtual multistack trees by dynamically selecting the least loaded available paths on the overlay network. The low computational complexity of the proposed algorithms leads to time and resource saving, as shown through extensive experiments. The Simulation results show that Network congestion avoidance strategy with optimal pacing scheme can efficiently improve the bandwidth utilization, Transmission Control Protocol (TCP) friendliness and reduce the packet drop rate in Pipeline Flux Leak Monitoring networks. Flood flow identified by the National Centre for testing: discussion group first proposed the use of particle filters to solve the new model can estimate the network congestion control problem. The results are sound, stable performance, efficiency 29%. Adaptive algorithm using the model proposed optimization scheme, to achieve accurate positioning of the leak, 0.05% measurement accuracy, positioning accuracy is improved 32%, more than 17% of the nodes in a more reliable routing path, reliable routing path of increase of 40%.

Index Terms—Optimal Pacing Scheme, Overlay networks, Congestion and contention, Particle filter, DIMRO, QoS

I. INTRODUCTION

Xinjiang, the country's oil, natural gas, liquefied petroleum gas rich areas, particularly the second phase in 2008 and 2007 natural gas to East Gas Pipeline Project implementation, in December 2009 China's first offshore natural gas pipeline - the West East Gas input to achieve second-line gas transport in Xinjiang. Under construction in Xinjiang Urumqi Petrochemical, Petrochemical alone, Junggar Basin, three extensions to ensure that the extension and the Second West-East Junggar Basin in Xinjiang completed simultaneously, while achieving gas transmission. Pipeline along the branch that needs a lot of intelligent instruments for flow measurement, control online distribution and pipeline leak detection, internal flow meter accurately measures the basic problem solving to achieve computer control. But the real-time online measurement and leak detection control dead zone exists, need to research and development. In particular, rely on measurement of the controller, temperature and pressure sensors on the instrument to achieve real-time online traffic and leak detection, the core issue is safety and Accurate location of leaks to ensure that pipeline flow interface, the branch office devices and other dangerous sites use security for open Internet. China's oil and natural gas reserves are mainly distributed in the western region, while the users are mainly distributed in central and eastern regions, pipeline transportation of its unique economy, convenience, security and other advantages are widely used in petroleum, natural gas transportation. Have the experts and scholars to conduct extensive research, but in real-time detection, reliability, accuracy and convenience of operation, several aspects of the comprehensive index, it is difficult to achieve a satisfactory level. How to achieve flow of the pipeline flow meter real-time online monitoring of leakage? How to solve the leaking pipe flow and accuracy of positioning accuracy, timeliness, ensure that the pipeline flow interface, the risk of branch offices and other sections of the security instrument used to achieve the open Internet? How the flow of oil and natural gas pipeline leak online monitoring is a key issue to be settled urgently. Optimal pacing scheme to solve flow pipeline leak location and monitoring to improve the positioning accuracy of the leakage point. The multi-hop backbone wireless network has to be utilized efficiently in order to improve the overall performance of the network. Hence all the packets going in a path have to be scheduled with an interval of FHD (Estimation of 4-hop transmission delay) to reduce the contention between the packets, thereby achieving better channel spatial reuse. [1] This spacing of packet transmissions is required in multi-hop wireless networks.
irrespective of higher layer protocols used when the flows are running for more than four hops. Our proposed Link Layer Adaptive Pacing (LLAP) scheme tries to reduce the contention in the network by properly scheduling the packets at edge nodes thereby increasing the channel spatial reuse in the network. We use a cross-layer approach for scheduling of packets and estimation of FHD in a path.

II. THE LINK LAYER ADAPTIVE PACING SCHEME MODEL AND FORMULATION

A. Model and hypothesis

Analysis of the use of intelligent algorithms to monitor the flow of leakage location model, implementation of innovation, first proposed the use of particle filters to estimate the new model to solve the network congestion control, design-based pipeline leak monitoring network traffic congestion control scheduling strategy in the fuzzy neural network model for scheduling Markov chain model and features, with the leakage flow into the pipeline network monitoring, adaptive real-time network transmission rate and reduce the network data loss rate, the model focuses on the intelligent algorithm combined with the leak to determine the problem, design flexibility in the use of TRIZ methods of the project solution. In order to implement the Differentiated Service (DiffServ) and at the core router dummy-net is used to emulate the network [2]. The experiments have been carried out using the following configuration:

(1) The packet size is fixed to 2000 bytes;
(2) Two colors were token bucket marker with a bucket size of 20000 bytes is used on the edge router;
(3) Routers are configured with a queue size of 100 packets and RIO parameters in the core router correspond to
   \[\text{weight} \times \text{rate} = (10, 20, 0.1, 20, 40, 0.02)\]
(4) The bottleneck between the core and the egress router has a fixed capacity of 2000 K bits/s;
(5) Measurements are carried out 20 times during 240 s for an FTP-like transfer.

We made experiments with a large set of different initial Round Trip Time delays and target rates. The choice of these results has been made since the various presented represent some of the worst cases for a unique flow (TCP and TFRG) to reach its target rate [3].

B. Node architecture and distributed scheduler

The implementation of Link Layer Adaptive Pacing (LLAP) scheme has two phases. One is the estimation of FHD on the path between each ingress and egress pair and the other one is sending packets into the network with pacing delay of FHD. [4] The FHD estimation on the path between ingress and egress pair is done by propagating the congestion information of the bottleneck node to the ingress node in a distributed manner. We provide a piece of node architecture and a distributed scheduler at the link layer for estimation of FHD for each ingress and egress pair. It is assumed that all core nodes in the backbone network maintain a separate queue called Input Queue for the packets destined to the same egress node and a Transmission Queue which contains the packets that are ready for transmission. The scheduler moves the packets from Input Queue to Transmission Queue. Both these queues serve the packets in First-In-First-Out (FIFO) fashion. All the incoming packets are placed into the corresponding Input Queue based on the egress node that a packet has to reach. This information will be obtained from the routing layer. Each node maintains the average time that the packets belonging to a particular egress node (d) D spent at this node (HT) and at the downstream node (NHT). The scheduler moves a packet from Input Queue to the Transmission Queue based on the values of HT and NHT. If NHT at a node d is greater than HT for egress node d, it adds additional delay to move the packet from its Input Queue to the Transmission Queue. At the ingress node the delay incurred in moving the packet from Input Queue to Transmission Queue is the estimated value of FHD for that egress node d. [5]

C. Propagation of congestion information

When ever a packet is transmitted from a node, the amount of time that the packet spent in this node is calculated by difference between the time of arrival of the packet to this node and the time of completion of packet transmission in the Medium Access Control (MAC) layer. Let us denote this by HTk,1, the estimation of HT at a node k. The amount of time the packet spent in the downstream node is the difference between the time at which the packet is transmitted from this node and the time at which the node overheard the transmission of the same packet from the downstream node. Let us denote this by NHTk,1, the estimation of NHT at a node k. We use a cross-layer approach for scheduling of packets and estimation of FHD for each egress node d. [6]

D. Particle filter algorithm

Suppose the dynamic and time-varying system model as follows: [6]

\[x_k = f(x_{k-1}, v_k), z_k = h(x_k, u_k)\]  (4)

In (4), the subscript k denotes the time index where \(K \in Z^+\), \(x_k\) is a state vector at the time of \(k\), \(v_k\) is process noise vector with independent and identical distribution, \(z_k\) is an observation vector at the time of \(k\), and \(n_k\) is Gaussian white noises with independent and identical distribution. If the initial probability is known as:

\[p(x_0 | z_0) = p(x_0)\]  (5)

Then, the prediction state equation can be written as:

\[p(x_k | z_{k-1}) = \int p(x_k | v_k)p(x_k | z_{k-1})dx_{k-1}\]  (6)

And the updating state equation can be written as:
\[ p(x_i | S_{i-1}) = \frac{p(z_i | x_i) p(x_i | S_{i-1})}{\sum_{x_i} p(z_i | x_i) p(x_i | S_{i-1})} \]  
(7)

Where
\[ p(z_i | x_i) = \int p(z_i | x_i, x_{i-1}) p(x_i | x_{i-1}) dx_i \]  
(8)

The importance function \( q(x_{i-1}, z_{i-1}) \) can be represented with continued product as:
\[ q(x_{i-1}, z_{i-1}) = q(x_{i-1}) \prod_{j=1}^{i-1} q(x_j | z_{0:j-1}, z_{i-1}) \]  
(9)

If we assume the state conform to Markov chains and the conditions of observation variables are independent and with given state, the Recursive formula about weight value can be obtained as follows:
\[ w_k = \frac{p(z_k | x_k) q(x_k)}{q(z_k | x_k, x_{k-1}) p(x_k | x_{k-1})} \]  
(10)

According to \( p(x_i | S_{i-1}) \), we can use the method of resampling to obtain \( N \) random sample points \( \{x_i^{(n)}\}_{n=1}^{N} \), so that the probability can be represented as:
\[ p(x_i | S_{i-1}) \approx \sum_{n=1}^{N} w_i^{(n)} \delta(x_i - x_i^{(n)}) \]  
(11)

The updating probability can be written as:
\[ p(x_i | z_{i-1}) \approx \sum_{n=1}^{N} w_i^{(n)} \delta(x_i - x_i^{(n)}) \]  
(12)

E. Pacing at the ingress node

Each upstream node in the path from the bottleneck node ensures that the amount of time of the packets belonging to a particular egress node spent in it is equal to the amount of time the packets spent at the bottleneck node. Now, the ingress node can estimate the FHD\( ^d \) by multiplying the estimate of \( NHT^d \) at this node by a constant \( k \) depending upon the hop length of the path. If the hop length is less than four then \( k \) is equal to hop length of the path, and four otherwise. Hence, the pacing delay at the ingress node for a given egress node \( d \) is calculated as follows: [7]
\[ PD^d = k \times NHT^d \]  
(13)

The scheduler moves the packets from the Input Queue to the Transmission Queue with a pacing delay (PD\( ^d \)) which makes the delay between successive transmission of packets for that particular egress node to be FHD\( ^d \).

F. DIMRO in QoS-aware overlay networks

Differentiated service Multicast algorithm for Internet Resource Optimization (DIMRO) algorithm is extended to leverage the differentiated service provided by QoS-aware overlay networks, i.e., able to provide differentiated services. Let us consider a multirate multicast scenario where receivers ask for the same data content but different rates and different QoS. In this scenario, the extended DIMRO algorithm allows a receiver to reuse the bandwidth already exploited by receivers asking for higher service classes without any extra cost for the network. By building a path from source \( s \) to a receiver \( r \), the DIMRO algorithm can reuse some sub-paths already exploited by higher service class receivers. Thus, receiver \( r \) obtains a better QoS, and the network saves resources because bandwidth already exploited by other receivers is reused for receiver \( r \). DIMRO in a QoS-aware overlay network proceeds into steps, described hereafter.

Step 0: Receivers are ordered according to their service class and rate. Let us consider a set \( R \) made up of \( M \) receivers. Let \( CL = \{c_1, \ldots, c_L\} \) be the set of service classes requested by receivers, where \( c_L \) is the highest service class and \( c_1 \) is the lowest one. The set of receivers \( R \) is partitioned into \( L = CL | \text{subsets}, [8] \)
\[ R = \bigcup_{i=1}^{L} R_i \]  
(14)

Subset \( R_i \) is made up of those receivers asking for service class \( c_i \), \( i = 1, \ldots, L \). Receivers in each subset \( R_i \) are ordered from the highest rate to the lowest one. Let us consider a partitioned and ordered set \( R \) made up of \( M \) receivers. Let \( r_j^i \) be the receiver \( j \) in the subset \( R_i \) with requested rate \( F_j^i \); \( |R_i| \) stands for the cardinality of \( R_i \), and \( i \) represents the associated service class.

Step i, \( i = 1, \ldots, L \): DIMRO connects source \( s \) with all those receivers asking for service class \( c_i \). Initially, the binary variables \( a_u \) are set to 1 for each overlay link \( (u, v) \). For receiver \( r_j^i, j = 1, \ldots, |R_i| \), \( a_u \) are set to 0 for all those overlay links \( (u, v) \) already used by receivers that ask for higher service class and lower or equal rate than \( r_j^i \). In fact, resources already exploited by the tree on these links could be used by \( r_j^i \) without any added cost.

Let \( b_u(c_i) \) be the available bandwidth of link \( (u, v) \) for the service class \( c_i \). A path \( p(s, r_j^i) \) from source \( s \) to a receiver \( r_j^i \) asking for cumulative rate \( F_j^i \) is feasible if \( b_u(c_i) \geq F_j^i \) for all its links. DIMRO chooses that feasible path \( p(s, r_j^i) \) from \( s \) to \( r_j^i \) that minimizes the following function, [9]
\[ f(p(s, r_j^i)) = \sum_{(u,v) \in p(s, r_j^i)} \alpha \cdot \exp\left(-b - \frac{m_{uv}(s, r_j^i)}{m_{uv}(s, r_j^i)}\right) \cdot \exp\left(-c \cdot \frac{\varepsilon}{\tau}\right) \]  
(15)

Where \( S_{\text{feas}}(s, r_j^i) \) is the set of feasible paths from \( s \) to \( r_j^i \). Coefficients \( a, b, \) and \( c \) has been determined by solving the following minimization problem of the quadratic error between the exponent model in (16) and the experimental results. Minimize:
\[ \sum_{i=1}^{L} \left[ a - \alpha \cdot \exp(-b \cdot \frac{m_{uv}(s, r_j^i)}{m_{uv}(s, r_j^i)}) \cdot \exp(-c \cdot \frac{\varepsilon}{\tau}) \right]^2 \]  
(17)

Starting from a set of chosen points
\[ \Gamma = \{(\varepsilon, \tau) : (s, r_j^i, a, b, c) \} \]  
(18)

The optimum exponent \( \alpha \) was experimentally obtained for each point \( (\varepsilon, \tau) \in \Gamma \) of the set. The coefficient has been chosen belonging to the interval \([0, 3]\) after an accurate tuning of the model. Values of coefficients \( a, b, c \) obtained by solving the minimization problem (17) are \( a = 3, b = 3.9, \) and \( c = 16.9 \). [10]
Exponent $\alpha$ is defined as in (16), and utilization $\rho_u(c_l)$ of the overlay link $(u, v)$ is calculated as

$$\rho_u = \frac{\rho_u(c_l) - \rho_u(c_l) - F_j}{\Delta}$$

(19)

Where $\rho_u(c_l)$ the bandwidth capacity of is overlay link $(u, v)$ for the class $c_l$, and $F_j$ is the cumulative rate requested by $r_j$.

For each overlay link $(u, v)$ belonging to path $p(s, r_j)$, if $a_{uv} = 0$ then the available bandwidth $b_{uv}(c_l)$ is not updated and the new path uses those resources already exploited by other receivers. Conversely, if $a_{uv} \neq 0$, then new resources have to be exploited and the new value of the available bandwidth becomes $b_{uv}(c_l) = b_{uv}(c_l) - F_j$

Then, the binary variable $a_{uv}$ is set to zero. [11]

The DIMRO algorithm determines the path $\overline{p(s, r_j)}$ by using a modified shortest-path algorithm. Common shortest-path algorithms could choose one of the possible paths with zero cost from $s$ to $u$, but this shortest path, as it will be clear in the following example, might not be correct. To determine a correct path the algorithm proceeds as it follows.

Starting from node $s$, the first link $(u, v)$ belonging to $p(s, r_j)$ with $a_{uv} \neq 0$ is found. Among all those paths in the tree exploiting a bandwidth equal or greater than $F_j$ and passing through node $u$, it is chosen the one that uses the highest service class. Let us indicate this path with $\overline{p_u}$. The new path $\overline{p(s, r_j)}$ from $s$ to $r_j$ is the concatenation of the sub-path from $s$ to node $u$, belonging to path $\overline{p_u}$, and the sub-path from node $u$ to $r_j$, belonging to $\overline{p(s, r_j)}$. [12]

Fig.1 shows a multicast tree on the overlay network where receiver $r_1$ (node 6) asks for a rate $F_1 = 2$ and a QoS mapped into service class $A$, receiver $r_2$ (node 8) asks for a rate $F_2 = 3$ and a QoS mapped into service class $B$, and receiver $r_3$ (node 7) asks for a rate $F_3 = 2$ and a QoS mapped into service class $C$. [13]

Indeed, in this case a discrepancy between the user and provider configurations either would induce a risk for the service user to get a poorer service than the negotiated one or for the service provider a risk to dedicate to the service user more resources than needed. We focus on the behaviors of QSTP on top of another network level QoS mechanism. This allows us to verify that the proposed protocol can be used over any kind of network providing a bandwidth guarantee. To perform this evaluation, we configure a QoS network with a Class Based Queuing (CBQ) scheduling mechanism that provides a guaranteed pipe of 300 K bit/s for the studied flow. The emulated QoS network does not use any admission control. The CBQ is configured in "borrow mode". It means that in case of non-congestion, the BE (best-effort) traffic can borrow bandwidth into the reserved pipe.
During the experiment, the UDP (User Datagram Protocol) flow emits at 200 K bit/s except between [60,120] s where it emits at 900 K bit/s. As a result, during this interval the bottleneck is full. Fig.2 (a) and (b) give the throughput measured at the sender and receiver side. Once the congestion occurs, the CBQ algorithm starts. Thanks to the CBQ scheduling, both flows obtain their guarantee as shown in these figures. Fig.2 (c) and (d) give the network level jitter in milliseconds obtained by both flows. We can see that TCP 2(c) obtains a higher jitter than QSTP 2(d). This is an expected result as TFRC (TCP Friendly Rate Control mechanism) congestion control algorithm has the property to emit non-burst traffic. In an obvious way, the resulting jitter must be lower.

**IV. SIMULATION RESULTS**

The increase in data rate of the source decreases the time interval between the consecutive packets from the source. As long as the interval between consecutive packets is less than the FHD, end-to-end delay will not change without applying LLAP. We can also see the same trend in the case of LLAP as LLAP will not incur artificial delay if the interval between consecutive packets is more than FHD. If the data rate of source increases further such that the interval between consecutive packets is less than FHD, the end-to-end delay increases with data rate and saturates. Without LLAP, the increase in end-to-end delay is due to the fact that, packet queuing delay and the contention delay in the path increases with data rate of the source. In the case of LLAP, the ingress node pushes packets into the path with interval of FHD between consecutive packets. So the increase in end-to-end delay is due to the packet queuing delay at the ingress node. The end-to-end delay saturates when the queue is full. This is shown in Fig. 3. From the figure, it is clear that introducing artificial delay at the ingress node does not increase the end-to-end delay of packets. The end-to-end delay is always less than the end-to-end delay when LLAP is not used.

We randomly picked five ingress and five egress nodes in each side of the grid and generated traffic between the ingress and egress nodes selected on opposite sides of the grid. We ran experiments with and without LLAP by varying the data rate of CBR flows. For packet size of 50 bytes, we varied the data rate of all CBR traffic from 1 Kbps to 16 Kbps. Fig. 4 shows the aggregate throughput for varying offered load. Without LLAP, as the offered load increases above 60 Kbps, the aggregate throughput decreases drastically.

In the experiment, by changing the length of the chain topology, we ran an FTP flow using TCP New-Reno...
connection between end nodes of the chain and measured the good-put with and without LLAP. For comparative purpose, we measured the performance of TCP with Adaptive Pacing (TCP-AP) without LLAP also. The results are shown in Fig. 5. The good-put of TCP New-Reno over IEEE 802.11 MAC without LLAP reduces drastically when the number of hops increases from 4 to 7. This is due to the fact that a number of hidden terminal collisions are higher near the source node. But LLAP reduces the contention in the path and improves the good-put of TCP New-Reno. We noted that TCP New-Reno with LLAP achieves good-put comparable to that of TCP-AP. As only one TCP flow exists in the chain topology, pacing at the transport layer spreads the packet transmission at the MAC layer as well. Here, we found that our LLAP scheme gives good-put comparable to that of TCP-AP. TCP-AP estimates the FHD using the Round Trip Time (RTT) and it works well only if both TCP source and sink are in WMN. But our LLAP scheme estimates the FHD in a distributed manner at the MAC layer and it works well even if one of them is in the Internet.

![Fig. 5 TCP good-put Vs hop length with and without LLAP](image1)

In source-specific communications, multicast sessions may have a large number of receivers with heterogeneous reception capacities. To accommodate this heterogeneity, we propose a novel layering scheme. In a layering scheme, data transmission through the network takes place over logical channels, i.e., a sender/receiver can simultaneously transmit/receive data on multiple channels. In a multirate multicast scenario, data on logical channel \( n \) is transmitted at rate \( w_n \). Receivers subscribe to the layers cumulatively, i.e., if a receiver subscribes to layer \( j \), it also subscribes to layers \( \{1,2,\cdots,j-1\} \) and receives data at the cumulative rate \( L_j = \sum_{i=1}^{j} w_i \).

Two of the most well-know multirate schemes are proposed. Channel rates are determined to minimize the total completion time, defined as the sum of all receivers’ completion time, i.e., the time that a receiver needs to download the file. With \( M \) receivers and \( K \) channels, the algorithm has a computational complexity \( O(M^3K) \). Channel rates are computed as follows: [14]

\[
W_j = \begin{cases} 
  b & \text{if } j = 1 \\
  \sum_{i=1}^{j-1} w_i & \text{if } j > 1 
\end{cases}
\]  

(20)

where \( b \) is a base rate. The layering scheme allows dynamic multicast groups, i.e., a receiver can join the group and start downloading the file at any time. This is possible since the transmission on each channel is cyclic, which in turn is possible because channel rates are given as in (20). However, the objective function of the layering scheme does not meet properties of fairness. We determine channel rates by minimizing the average ratio between requested rate and cumulative rate subscribed by a receiver. We formulate hereafter the channel rate assignment problem, while we refer the reader to the Appendix for the detailed description of an efficient algorithm to solve this problem in polynomial time. This layering scheme is exploited by our overlay multicast algorithms.

Fig. 6 shows that the DIMRO Rejection Rate in Network1 is lower than the Rejection Rate of the Optimal solution of the Steiner Tree Problem (OSTP) with \( c_{nw} = 1 \). The Rejection Rate is approximately the same until the Number of Requested Trees is less than 2500. When the Number of Requested Trees overcomes this threshold, DIMRO shows a lower Rejection Rate.

![Fig. 6. DIMRO and OSTP Rejection Rate in Network](image2)

Both DIMRO and OSTP Network Load in Network1. Both DIMRO and OSTP Rejection Rates grow at the growing of the Number of Requested Trees because the same bottlenecks occur. These bottlenecks depend on the network topology and cannot be avoided, but the DIMRO Rejection Rate is lower because bottlenecks occur later. Fig. 7 shows that the DIMRO Network Load is lightly lower than the OSTP Network Load until the Rejection Rate is the same. Then, since DIMRO rejects fewer trees, its Network Load is higher than the OSTP one. DIMRO and the optimal solution of the Steiner tree problem with unitary costs are compared by using two metrics: the Rejection Rate and the Network Load. For each simulation campaign several experiments have been run to ensure 95\% relative confidence intervals smaller than 5\%. Multicast groups are sequentially randomly generated. Multicast group members (source and receivers) are randomly chosen among network nodes. The number of receivers for each multicast group is uniformly distributed from 5 to 15, and the bandwidth request of each receiver is uniformly distributed from 0.1
to 2 Mbps. Flood flow identified by the National Centre for testing: discussion group first proposed the use of particle filters to solve the new model can estimate the network congestion control problem. The results are sound, stable performance, efficiency 29%. Adaptive algorithm using the model proposed optimization scheme, to achieve accurate positioning of the leak, 0.05% measurement accuracy, positioning accuracy is improved 32%, more than 17% of the nodes in a more reliable routing path, reliable routing path of increase of 40%.[15]

V. CONCLUSIONS

It proposed a Link Layer Adaptive Pacing scheme to control the flow of packets into the backbone network at all ingress nodes in order to effectively utilize the capacity of the backbone network. The available capacity of each path between the ingress and egress pair of nodes is found using a distributed way of estimating the four hop transmission time on a multi-hop path. We found that LLAP improves the performance of both reliable and unreliable transport layer protocols such as TCP and UDP, respectively in different network scenarios. Their objective is to achieve traffic balancing on the overlay network in such a way as to avoid traffic congestion and fluctuation in the underlay network, which cause low performance. To address these problems, the algorithms actively probe the underlay network and compute virtual multicast trees by dynamically selecting the least loaded available paths on the overlay network. At home and abroad in the pipeline leak monitoring system for the successful application of all aspects of rare, although there are some more mature foreign technology, but most do not apply the actual situation of China's pipeline[16]. Kaifeng Instrument Co., Ltd. R & D in 2008 two-way dual-ball-style volume control is the issue of the use of Zhou Peng immune ant colony algorithm design intelligent genetic algorithm combined with the leak problem, the establishment of channels of the mathematical model for leak location, to achieve fast and accurate positioning of key technologies application of the results, products, accuracy of 0.05%, to fill the gaps in the technology internally. [7-9] West-East Gas Transmission Project Simple devices can be for cutting plate and the use of leakage alarm switch online positioning of new technologies, widely used in petroleum, chemical, and gas flow control industries. Ventura flow meter instruments are mainly used in Shenzhen to Hong Kong's water supply system, the use of the first issue of the use of particle filters to estimate the new model to solve the network congestion control, design-based pipeline leak monitoring network traffic congestion control in key technologies, such as scheduling strategy the application of the results. The results of the project to directly serve the project and the Sichuan Gas Project in the East lost oil, natural gas pipeline flow measurement and online monitoring of leakage, the results can be technology promotion and achievements, be directly applied to the petrochemical enterprises in Xinjiang pipeline flow measurement and online monitoring of leakage, Xinjiang the sustainable development of local industry to provide technical support and exemplary role. Zhou Peng presided over issues, "the flow of intelligent devices Leakage Detection and Simulation Technology" on intelligent algorithm has been applied to the actual development of products to generate significant economic and social benefits, and in 2009 won the school science and technology II prize.

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ZHOU Peng (1970-), Male (Han), Henan Province, received the BS degree in Electronic and Information Engineering from Information Engineering University in 1993, and the MS degree in Control Theory and Control Engineering from Northwestern Polytechnical University in 2008. He has been with the College of Information Engineering, Tarim University at alar xinjiang, since 1999, where he is currently an Adjunct Professor. His current research interests include computer applications, electrical automation, and communications engineering and electronic technology.