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A Framework to avoid Video Distortion in Wireless Multihop Networks

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***Abstract*-Typical routing algorithms designed for wireless network are application agnostic, so to overcome this we consider a wireless network where the application flows consists of video traffic. Reducing this distortion is critical for the user. Using link quality based routing metrics cannot minimize video distortion. So, we construct an analytical framework to understand first and then to assess the impact of wireless network on video distortion. Using this we can formulate a routing policy for minimizing distortion. We find via experiments that our protocol is efficient in reducing video distortion.**

Keywords-Wireless network, Protocol, Distortion, Routing, Analytical Framework.

I. INTRODUCTION

Video traffic has become a problem nowadays due to the increase in the use of wireless networks. Maintaining a good quality of video is very important. The video quality is affected by: 1) the distortion due to compression at the source and 2) distortion due to both wireless channel induced errors and interference.

Groups like I, P and Btype frames provide different levels of encoding. In I frame information is encoded independently, in P and B frames information is encoded relative to information encoded within other frame.

Video quality can be improved by accounting for application requirements. The schemes used to encode a video clip can accommodate a certain number of packet losses per frame. If the number of lost packets exceeds a threshold value then the frame cannot be decoded correctly. Thus, resulting a distortion. The value of distortion depends on position of unrecoverablevideo frames in the GOP (Group of Pictures). So, we construct an analytical model to view the behaviour of the process that describes the evolution of frame losses in the GOP. Using this we capture how the choice of path for an end-to-end flow affect the performance of a flow in terms of video distortion.

Our model is built based on a multilayer on approach as shown in fig1.The packet-loss probability on a link is mapped to the probability of a frame loss in the GOP and the frame loss probability is then directly associated with the video distortion

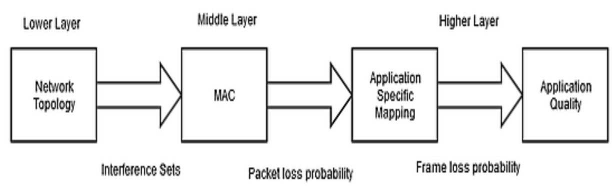

Fig1 : System Architecture

Metric.Using the above mapping from the network-specific property to the application-specific quality metric, we indicate the problem of routing as an optimization problem where we can find the path from the source to the destination that can minimizes the end-to-end distortion.

The solution for this problem is based on a dynamic programming approach that effectively captures the evolution of the frame –loss. After this we design a practical routing protocol, based on the above solution, to minimize routing distortion.

II. LITERATURE SURVEY

1) Overview of the H.264/AVC video coding standard:

H.264/AVC is newest video coding standard of the ITU-T Video Coding Experts Group and the ISO/IEC Moving Picture Experts Group. The main goals of the

H.264/AVC standardization effort have been enhanced compression performance and provision of a "network-friendly" video representation addressing "conversational" (video telephony) and "nonconversational" (storage, broadcast, or streaming) applications[1][2]. H.264/AVC has achieved a significant improvement in rate-distortion efficiency relative to existing standards. This article provides an overview of the technical features of H.264/AVC, describes profiles and applications for the standard, and outlines the history of the standardization process.

2)A high throughput path metric for multi-hop wireless routing:

This paper presents the expected transmission count metric (ETX), which finds high-throughput paths on multi-hop wireless networks. ETX minimizes the expected total number of packet transmissions (including retransmissions) required to successfully deliver a packet to the ultimate destination. The ETX metric incorporates the effects of link loss ratios, asymmetry in the loss ratios between the two directions of each link, and interference among the successive links of a path[3][4]. In contrast, the minimum hop-count metric chooses arbitrarily among the different paths of the same minimum length, regardless of the often large differences in throughput among those paths, and ignoring the possibility that a longer path might offer higher throughput.This paper describes the design and implementation of ETX as a metric for the DSDV and DSR routing protocols, as well as modifications to DSDV and DSR which allow them to use ETX. Measurements taken from a 29-node 802.11b test-bed demonstrate the poor performance of minimum hop-count, illustrate the causes of that poor performance, and confirm that ETX improves performance. For long paths the throughput improvement is often a factor of two or more, suggesting that ETX will become more useful as networks grow larger and paths become longer.

3) Packet loss resilient transmission of MPEG video over the internet

A method is proposed to protect MPEG video quality from packet loss for real-time transmission over the Internet. Because MPEG uses inter-frame coding, relatively small packet loss rates in IP transmission can dramatically reduce the quality of the received MPEG video. In the proposed high-priority protection (HiPP) method, the MPEG video stream is split into high- and low-priority partitions, using a technique similar to MPEG-2 data partitioning. Overhead resilient data for the MPEG video stream is created by applying forward error correction coding to only the high-priority portion of the video stream. The high- and low-priority data, and resilient data, are sent over a single channel, using a packetization method that maximizes resistance to burst losses, while minimizing delay and overhead. Because the proposed method has low delay and does not require re-transmission, it is well suited for interactive and multicast applications[5][6]. Simulations were performed comparing the improvement in video quality using the HiPP method, using experimental Internet packet loss traces with loss rates in the range of 0–8.5%. Overhead resiliency data rates of 0%, 12.5%, 25%, and 37.5% were studied, with different compositions of the overhead data for the 25% and 37.5% overhead rates, in an attempt to find the "best" composition of the overhead data. In the presence of packet loss, the received video quality, as measured by PSNR and the Negsob measure, was significantly improved when the HiPP method was applied [10][11].

III. EXISTING SYSTEM

The encoding and transmission of video indicates the significance of video communication. Different approaches exist in handling such an encoding and transmission. Multiple description coding is a coding technique that fragments a single media stream into n sub streamsreferred to a description. The packets of each description are routed over multiple (partially), disjoint paths. In order to decode the media stream any description can be used. The idea of MDC is to provide error resilience to media stream.

Layered coding (LC) mechanism generates a base layer and n enhancement layers. The base layer is necessary for the media stream to be decoded, enhancement layers are applied to improve stream quality.The first enhancement layer depends on the base layer and each enhancement layer n+1 depends on its sub-ordinates layer n.

We use standards like the MPEG-4 and the H.264/AVC which provide guidelines for a transmission over a communication system based on layer coding.The initial video clip is separated into a sequence of frames of different importance with respect to quality and hence different levels of

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encoding. The frames are called I-,P-,B-frames constitute a structure named the GOP.

In another existing model, an analytical framework is developed to model the effects of wireless channel fading on video distortion. In other existing model, the authors examine the effect of packet-loss patterns and specifically the length of error bursts on the distortion of compressed video.

IV. MODEL FORMULATION

Initially the analytical model joins the functionality of the physical layer and MAC layer with the application layer to send the video from source to the destination. The position of the first unrecoverable frame in the GOP gives the value of the distortion.

A. Physical and MAC layer Modeling

Considering an IEEE 802.11 network where the set of nodes is denoted by N .Since this model is application agnostic this provides the packet loss probability due to traffic and interference in the network.

Using the Network loss model we derive the 3 equations. the first is to scheduling model, that computes the serving rate Pi, p of a path at each node, as a function of the scheduler coefficient Ki, p and the service time.

$$Pi,p=Ki,pE[Ti,p] \qquad (1)$$

The second captures the MAC and PHY and associates the probability βi,p of a failure with access probability ói,p

$$ói,p= \frac{2(1-2βi,p)}{W(1-2Bi,p)+βi,p(w+1)(1-(2βi,p)B)} \qquad (2)$$

Where B is the number of back off stages and W is the minimum window size. The third describes the routing model and computes the incoming traffic rate λj,p

$$λj,p=KI,P(1-Bi,p) \quad (3)$$

Using the iteration we join the equations until the result is achieved. Which gives the approximate packet loss probability.

B. Video distortion model

Since we are using the multi-hop we, develop a analysis model where it captures the evaluation of the distortion at different links from source to destination.by considering a GOP structure which consists of an I-frame followed by P-frames. So the index each frame in the GOP starting from 0, i.e., the I-frame corresponds to index 0, and the P-frames correspond to indices from 1 up to. We focus on predictive source coding where, if the ith frame is the first lost frame in a GOP, then the ith frame and all its successors in the GOP are replaced by the (i-1) st frame at the destination node.

The average distortion D(i) is computed using linear model which gives the average mean square error is:

$$D^{(i)}=(F-i)*\frac{i*F*Dmin+(F-i-1)*Dmax}{(F-1)*F} \qquad (4)$$

For i=0,1,2,3---------(F-1)

$Dmin=D^{(F-1)}$ is achieved when the last frame of GOP is lost.$Dmax=D^{(0)}$ is achieved when the first frame is lost.We can define the sequence D={Dt,t=0,1,2} to represent the wireless transmission distortion along the path from the source to the destination, where Dt is the wirelesstransmission video distortion at the t th hop. So, the distortion can be one of the following:

$$\{D^{(0)}, D^{(1)},---------D^{(F-1)}\}U\{0\} \quad (5)$$

The number of packet losses per frame can be computed by defining the multi- dimensional process M={Mt,t=0,1,2,---}

$$Mt=(Mt^{(0)},Mt^{(1)},Mt^{(F-1)}) \qquad (6)$$

If the β is the packet loss probability then the transmission probability of process M at hop t is: $Mt=(i_0,i_{1,}i_2,-----i_{(F-1)})$

At hop t+1 is:$M_{t+1}= (j_{0,}\ j1, ----j_{\ (F-1)})$ (7)

The transition probability of first component $M^{(0)}$ is

$$Φ^{(0)}_{i0j0} =P\{Mt_{+1}{}^{(0)}=j0|Mt^{(0)}=io\} \qquad (8)$$

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Volume 5, Issue 5, May 2016

C. Video distortion dynamics

The position of the first unrecoverable frame in the GOP gives the value of the distortion D at hop t from source to the destination. The process C can be defined as:

$$C = \{C_t,\ t=0,1,2, \text{----------}\} \quad (9)$$

The value of the process at set is:

$$C = \{0, 1, 2, \text{-----}F-1, F\} \quad (10)$$

The value 0 indicates that the I frame is lost, the value between the 1 and F-1 indicatesthat the P frame is lost. And the value F indicates that no frame is lost that is the value of distortion is 0.The transition probabilities at hop t=0, 1, 2. --- Of the process C

$$P_t(I,j)=P\{C_{t+1}=j \mid C_t=i\} \quad (11)$$

The value of the video transmission distortion depends on the value of the process at hop t is:

$$D_t = \begin{cases} 0, & \text{if } C_t=F \\ D^{(c)}, & \text{if } C_t=c \text{ and } 0<=c<=(F-1) \end{cases} \quad (12)$$

V. PROTOCOL DESIGN

The solution to the MDR problem can be computed by using the source node. The source node will sample the network to gather the information about the state of the network during the path discovery process using which the ETX can be computed which measures the quality of the network.

After this estimation the “Route Request” message is passed during the Route Discovery phase. After receiving the “Route Request” message the Algorithm 1 steps are followed, which defines the initial state as x-(s, F), where F is the GOP size. The boundary size is defined as B, which represents the terminating set for the optimization process.

Next Algorithm 2 is called which produces the next node in the path.

Algorithm 1: Path discovery (Uses Algorithm 2)

Input: source node s, destination node d.

Input: Frame size F.

Output: route R from s to d.

/* Route discovery */

send Route Request

receive Route Replay (n_i,ETX_i) message

N= {node-ids n_i from Route Reply messages}

/* path discovery */

n<- S

c<- F

B= {(d, c) | 0<=C<=F}

R<- [ ]

x<- (n, c)

append x to R

/* Path computation /*

repeat

u* <- next_node_in_optimal path(x,B,N)

c^ <- E[Cnew | Ccur=c]

n<- u*

c <-c^

x <- (n, c)

append x to R

N<- N-{u*}

until xϵ B

Algorithm 2: Next Node in optimal path

Input: Initial state $x_{s,}$ boundary set B

Input: set of available nodes N

Output: net node n* in the optimal path

/* Initialization */

C= {0, 1,----F}

L=N*C

T <- ||L||

/* optimal control computation */

for i=T to 1 do

if i=T then

for all xϵH do

Ji(x) <- K(x)

end for

else

for all x= (n, c) ϵ H do

U (n) <- {n` | n, n`1-hop neighbors}

j_i(x, u) <- {g(x, u) + sum of x` $P_i$ (c, c` | 0) J_{i+1} (x`)}

J_i(x) <- $\min_{u \epsilon U}$(n) j_i(x, u)

Pi(x) <- arg min uϵU (n) j_i(x, u)

end for

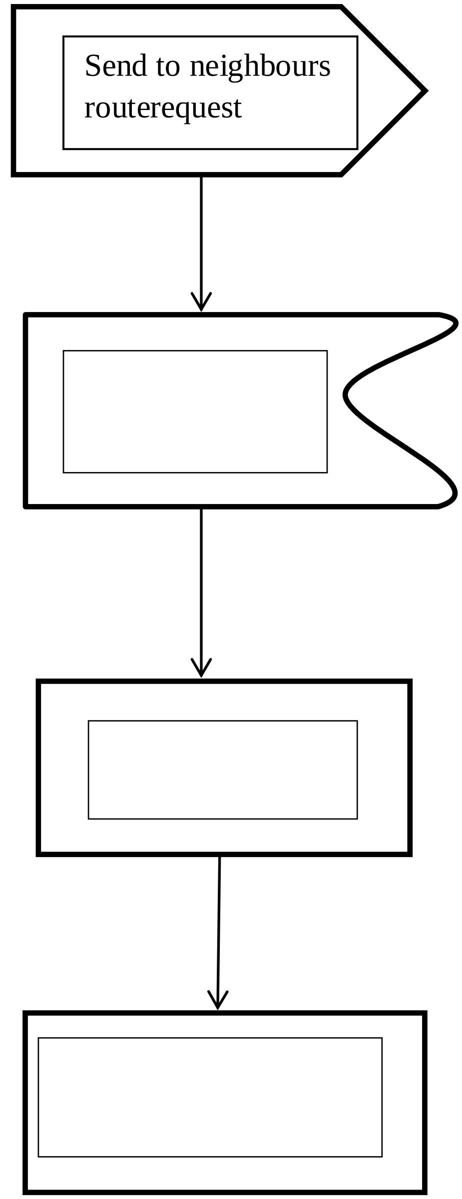
end if

end for

n* <- p, (xs)

return n*

The operation of the source node can be depicted by the flow chart as follows:



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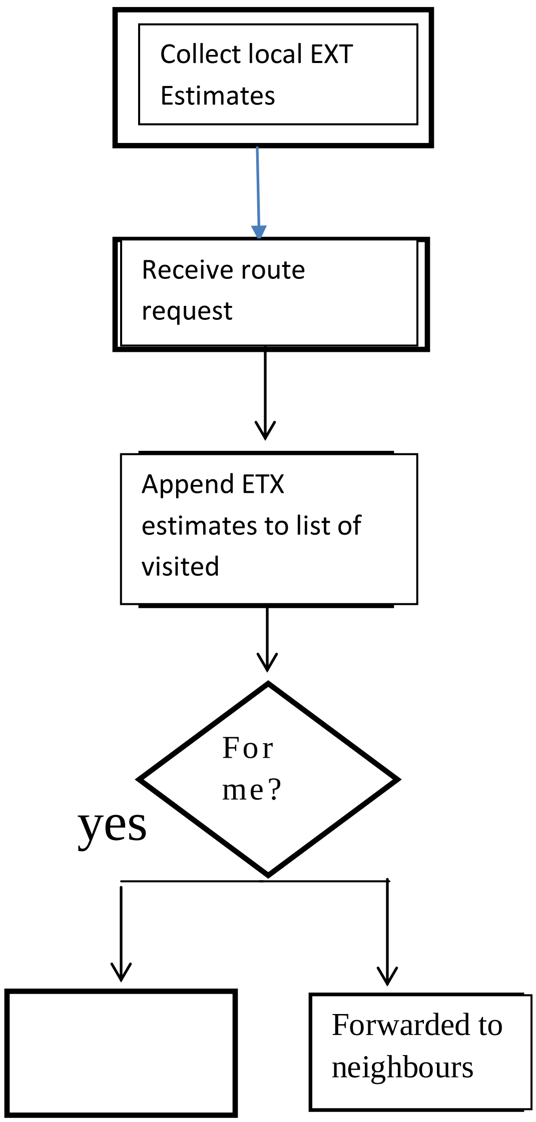
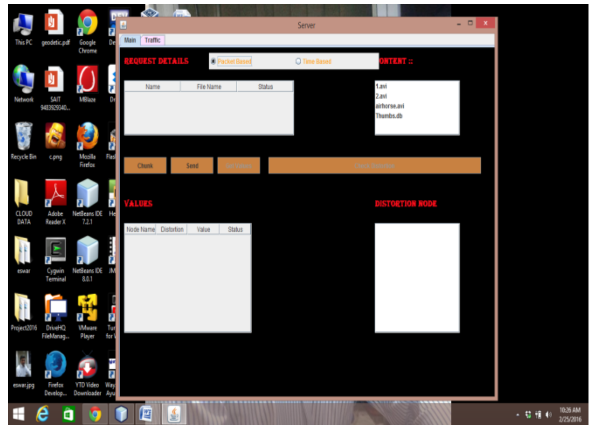
Flow chart for intermediate node and destination node:

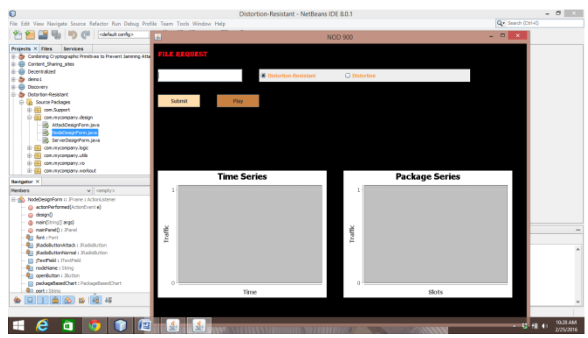


VI. IMPEMENTATION RESULTS

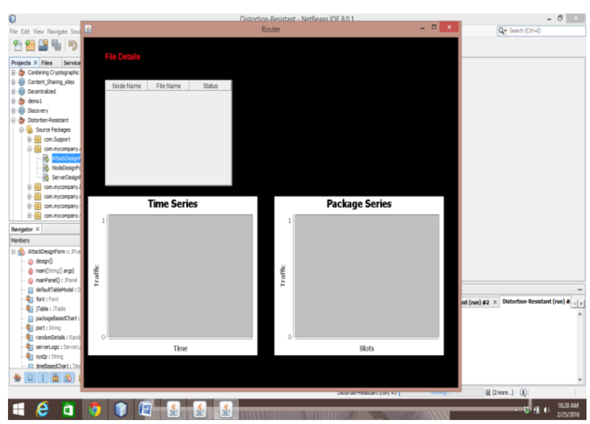
Server displaying the Request details:



Router:

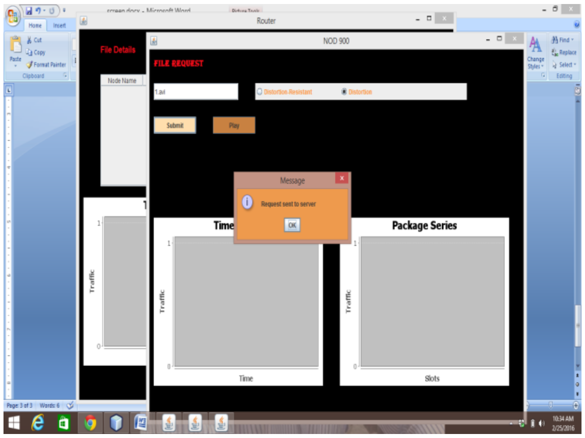


Client:

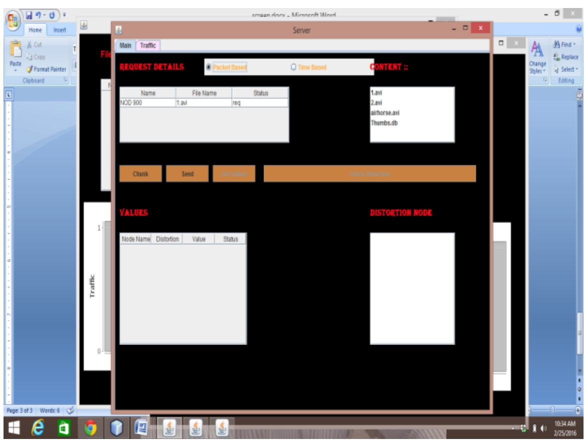


ISSN: 2278 – 1323
International Journal of Advanced Research in Computer Engineering & Technology (IJARCET)
Volume 5, Issue 5, May 2016

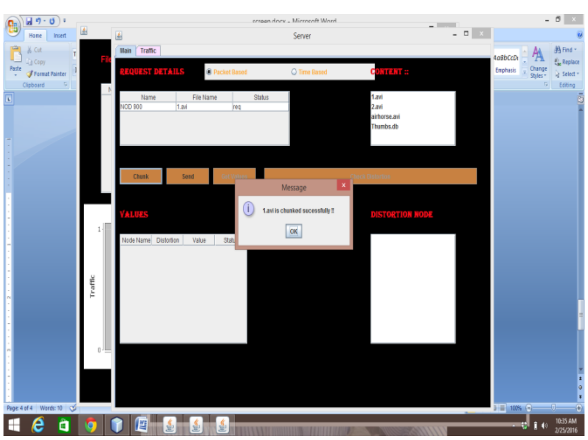
Demonstration of distraction:



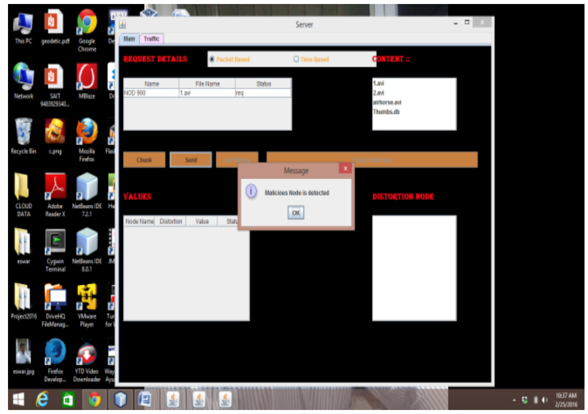
Request received from client:



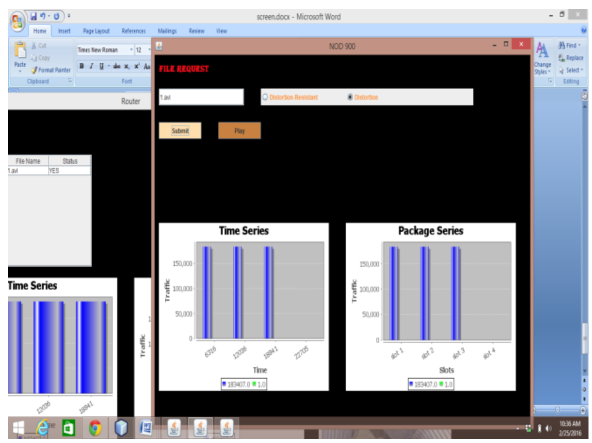
Encoding done on server:



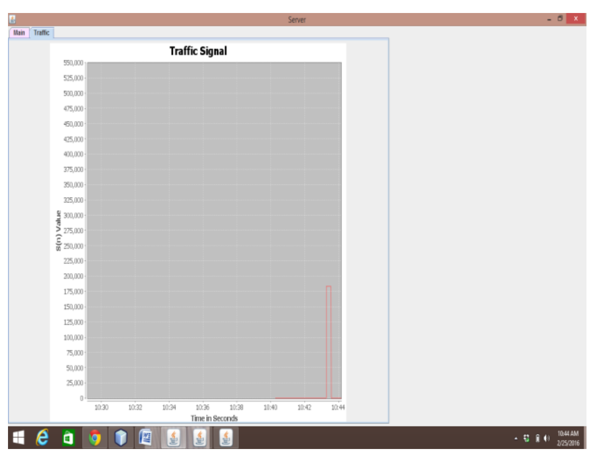
Attacker detected in path:



Packet all at receiver:



Traffic signal:



VII. CONCLUSION

The routing policy is application-aware that provides benefits in terms of user-perceived performance. Specifically, we consider a network that primarily carries video flows. The impact of routing will be on end-to-end distortion of video flows. For this, we construct an analytical model that ties video distortion to the underlying packet-loss probabilities. Using this model, we find the optimal route (in terms of distortion) between a source and a destination node using a dynamic programming approach. Based on our approach, we design a practical routing scheme that we then evaluate via extensive simulations and test bed experiments. Traditional metrics such as ETX, our approach takes into account correlation across packet losses that influence video distortion. Our simulation study shows that the distortion is decreased by 20% compared to ETX-based routing. The user experience degradation due to increased traffic load in the network is kept to a minimum.

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