Performance of Traffic Content Patterns Streaming Using Scalable Transcoding Controller
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Abstract:
Due to the increasing quality of multimedia system streaming applications and services in recent years, the problem of trusty video delivery to stop undesirable content-leakage has, indeed, become vital. Whereas protective user privacy standard systems has self-addressed this issue by proposing strategies supported the observation of streamed traffic of the network, these conventional systems maintain a high detection accuracy whereas managing a number of the traffic variation within the network however their detection performance considerably degrades as a result of the numerous variation of video lengths. In this paper, we have a tendency to specialize in overcoming this issue by proposing a completely unique content-leakage detection theme that's strong to the variation of the video length. By comparison videos of various lengths, we have a tendency to verify a relation between the length of videos to be compared and the similarity between the compared videos. Therefore, we have a tendency to enhance the detection performance of the projected theme in AN environment subjected to variation long of video. Through effectiveness of our projected theme is evaluated on variation of video length, delay variation, and packet loss.

Introduction
In recent years, with the speedy development of broadband technologies and therefore the advancement of high-speed wired/wireless networks, the recognition of period of time video streaming applications and services over the web has enlarged by leaps and bounds. YouTube and Microsoft network video area unit notable samples of such applications. They serve a large population of users from all round the world with various contents, starting from daily news breeds to diversion feeds as well as music, videos, sports, then forth, by exploitation streaming transmission technologies. Additionally, period of time video streaming communications like internet conference in intercompany networks or via net with virtual personal networks (VPNs) area unit being widely deployed during a larger range of firms as a robust means that of expeditiously promoting business activities while not further prices. An important concern in video streaming services is that the protection of the bit stream from unauthorized use, duplication and distribution. One in all the foremost standard approaches to stop undesirable contents distribution to unauthorized users and/or to guard authors’ copyrights is that the digital rights management (DRM) technology. Most DRM techniques use cryptanalytic or digital watermark techniques but, this type of approaches don't have any vital impact on distribution of contents, decrypted or reconditioned at the user-side by licensed however malicious users. Moreover, distribution is technically now not troublesome by exploitation peer-to-peer (P2P) streaming code. Hence, streaming traffic could also be leaked to P2P networks.

Literature Survey
1) Performance Analysis of IP Security VPN
This web Protocol Security (IPSec) could be a protocol suite for securing web Protocol (IP) communications by authenticating and encrypting every IP packet of an information stream. IPSec design needs the host to produce confidentiality mistreatment Encapsulating Security Payload and knowledge integrity mistreatment either Authentication Header or Encapsulating Security Payload and anti-replay protection. IPSec has become the foremost common network layer security management and, a wide deployed mechanism for implementing Virtual personal Networks (VPNs). This paper presents association analysis of IPSec VPN for videoconference in real time traffic over a secure communication links by implementing an IPSec-based VPN technology.

2) Dynamic Scalable Model for Video Conferencing (DS MVC) Using Request Routing
The Session Initiation Protocol (SIP) is associate application-layer management protocol that is employed to form, alter and stop transmission sessions. These transmission sessions embody decision and video conferences, net telephone and alternative such events. In general, SIP has centralized videoconference system, which faces the drawbacks of quantifiability and information measure. Once there’s a requirement to own a transmission conference on a large-scale, the problem of excessive load on Conference Server (CS) should be resolved. During this paper we have a tendency to alter this bottleneck issue on
Conference Server by examining the shortcomings of current transmission conference model. We have a tendency to propose a dynamic scalable model for SIP-based conferences DSMVC (Dynamic scalable Model for Video Conferencing), which is able to facilitate North American nation reducing the traffic from metallic element. During this model, purchasers act as a metallic element instead of employing a one metallic element for each consumer. As a result, once the amount of purchasers will increase, the amount of metallic element will increase consequently.

3) Advances in Digital Video Content Protection

Illegal copying and distribution on a massive scale facilitates misuse, piracy, plagiarism and misappropriation. Example n Peer-to-peer (P2P) architecture based systems create considerable challenges for copyright enforcement. P2P with anonymity, DRM for P2P. The use of DRM is controversial. Usage rules of DRM system may contravene the rights and privileges granted to the public under copyright law. Information releasing may violate expectations of privacy. DRM may reduce competition. DRM may increase cost of devices.

4) Enabling Conferencing Applications on the Internet using an Overlay Multicast Architecture

In response to the serious scalability and deployment concerns with IP Multicast, we and other researchers have advocated an alternate architecture for supporting group communication applications over the Internet where all multicast functionality is pushed to the edge. We refer to such architecture as End System Multicast. While End System Multicast has several potential advantages, a key concern is the performance penalty associated with such a design. While preliminary simulation results conducted in static environments are promising, they have yet to consider the challenging performance requirements of real world applications in a dynamic and heterogeneous Internet environment. In this paper, we explore how Internet environments and application requirements can influence End System Multicast design. We explore these issues in the context of audio and video conferencing: an important class of applications with stringent performance requirements. We conduct an extensive evaluation study of schemes for constructing overlay networks on a wide-area test-bed of about twenty hosts distributed around the Internet. Our results demonstrate that it is important to adapt to both latency and bandwidth while constructing overlays optimized for conferencing applications.

Further, when relatively simple techniques are incorporated into current self-organizing protocols to enable dynamic adaptation to latency and bandwidth, the performance benefits are significant. Our results indicate that End System Multicast is a promising architecture for enabling performance-demanding conferencing applications in a dynamic and heterogeneous Internet environment.

EXISTING SYSTEM

- A DRM (Digital Rights Management) technology which includes cryptographic or watermark techniques is used.
- It maintains high detection accuracy while copying with some of the traffic variation in the network.
- It is difficult to entirely prevent streaming content leakage by means of packet filtering.
- Video of different length in the network environment causes a considerable degradation in the leakage detection performance.

DISADVANTAGES

- It is difficult to protect the bit stream from unauthorized use, duplication and distribution.
- Redistribution is technically no longer difficult by using peer-to-peer (P2P) streaming software. Hence streaming traffic may be leaked to P2P networks.
- Detection performance substantially degrades owing to the significant variation of video lengths.

PROPOSED SYSTEM

- The proposed system provided an efficient interactive streaming service for diversified mobile devices and dynamic network environments.
- When a mobile device requests a multimedia streaming service, it transmits its hardware and network environment parameters to the profile agent in the cloud environment, which records the mobile device codes and determines the required parameters.
- Then transmits them to the Network and Device-Aware Multi-layer Management (NDAMM). The NDAMM determines the most suitable SVC code for the device according to the parameters, and then the SVC Transcoding Controller (STC) hands over the transcoding work via map-reduce to the cloud, in order to increase the transcoding rate.
- The multimedia video file is transmitted to the mobile device through the service.

ADVANTAGES

- The network bandwidth can be changed dynamically.
- This method could provide efficient self-adaptive multimedia streaming services.
SYSTEM ARCHITECTURE

User Profile Module:
The profile agent is used to receive the mobile hardware environment parameters and create a user profile. The mobile device transmits its hardware specifications in XML-schema format to the profile agent in the cloud server. The XML-schema is metadata, which is mainly semantic and assists in describing the data format of the file. The metadata enables non-owner users to see information about the files, and its structure is extensible. However, any mobile device that is using this cloud service for the first time will be unable to provide such a profile, so there shall be an additional profile examination to provide the test performance of the mobile device and sample relevant information. Through this function, the mobile device can generate an XML-schema profile and transmit it to the profile agent. The profile agent determines the required parameters for the XML-schema and creates a user profile, and then transmits the profile to the DAMM for identification.

1. Network and Device Aware Multi-Layer Management (NDAMM):
The NDAMM aims to determine the interactive communication frequency and the SVC multimedia file coding parameters according to the parameters of the mobile device. It hands these over to the STC for transcoding control, so as to reduce the communication bandwidth requirements and meet the mobile device user's demand for multimedia streaming. It consists of a listen module, a parameter profile module, a network estimation module, a device-aware Bayesian prediction module, and adaptive multi-layer selection. The interactive multimedia streaming service must receive the user profile of the mobile device instantly through the listen module. The parameter profile module records the user profile and determines the parameter. This is provided to both the network estimation module and the device-aware Bayesian prediction module to predict the required numerical values. Rw and Rh represent the width and height of the supportable resolution for the device, CPavg and CP represent the present and average CPU operating speed. Db and Db rate represent the existing energy of the mobile device and energy consumption rate, and BW, BWavg, and BWstd represent the existing, average and standard deviation values of the bandwidth. When this parameter form is maintained, the parameters can be transmitted to the network estimation module and the device-aware Bayesian prediction module for relevant prediction.

2. Dynamic Network Estimation Module (DNEM):
The DNEM is mainly based on the measurement-based prediction concept; however, it further develops the Exponentially Weighted Moving Average (EWMA). The EWMA uses the weights of the historical data and the current observed value to calculate gentle and flexible network bandwidth data for the dynamic adjustment of weights. In order to determine the precise network bandwidth value, the EWMA filter estimates the network bandwidth value, which is the estimated bandwidth of the No. t time interval, is the bandwidth of the No. time interval, and is the estimation difference. For different mobile network estimations, this study considered the error correction of estimation and the overall standard difference and estimated the different bandwidths by adjusting the weights among which, is the moving average weight and is the standard deviation weight. When the prediction error is greater than, the system shall reduce the weight modification of the predicted difference; relatively, when the prediction error is less than, the system shall strengthen the weight modification of the predicted difference. When the changed bandwidth of the system is greater than the standard difference, the predicted weight will increase as the corrected value of the standard deviation is reduced. The predictor formula for the overall mobile network quality uses the standard normal state value range concept of plus-minus three standard deviations of statistics, referring to identify the stable or unstable state of the current mobile network.

http://ijesc.org/
3. Network and Device-Aware Bayesian Prediction Module (NDABPM):
The SVC hierarchical structure provides scalability of the temporal, spatial and quality dimensions. It adjusts along with the FPS, resolution and video variations of a streaming bitrate; however, the question remains of how to choose an appropriate video format according to the available resources of various devices. Hereby, in order to conform to the real-time requirements of mobile multimedia, this study adopted Bayesian theory to infer whether the video features conformed to the decoding action. The inference module was based on the following two conditions:

- The energy of the mobile device shall be sufficient for playing a full multimedia video. Full multimedia service must be able to last until the user is satisfied. This assumed condition is also the next main decision rule. As for the three video parameters of FPS, resolution and bit rate, the bit rate depends on the frame rate and resolution, so the Bayesian network adopts the frame rate and resolution as the video input features and uses the bit rate as parameter considered.

4) Proposed Adaptive Communication and Multi-Layer Content Selection (ACMCS):
When the predicted bandwidth state and the Bayesian predictive network are determined, the cloud system will further determine the communication and the required multimedia video files according to the information.

Communication Decision:
A good dynamic communication mechanism can reduce the bandwidth needs and the power consumption of the device resulting from excessive packet transmission, and the transmission frequency can be determined according to the bandwidth and its fluctuation ratio based on such dynamic decision-making. The transmit mode is engaged until the device finds a variation of the transmitted variables that exceeds a threshold. Although the threshold can reduce the communication frequency effectively and precisely, in this mode the mobile device must start up additional threads for continuous monitoring; thus, the load on the device side is increased. When the network bandwidth difference exceeds a triple standard deviation, this indicates the present network is unstable. The overall communication frequency shall incline to frequency to avoid errors; however, when the network bandwidth difference is less than a triple standard deviation, the current network is still in a stable state, and the influence on bandwidth difference can be corrected gradually. Profile examination and subsequent hardware features. When the Bayesian inference table is completed, the next communication time can be determined, and the SVC multimedia coding applicable for the mobile device can be provided according to the predicted and inferred network and hardware features.

Conclusion
For mobile multimedia streaming services, how to provide appropriate multimedia files according to the network and hardware devices is an interesting subject. In this study, a set of adaptive networks and a device aware QoS approach for interactive mobile streaming was proposed. The DNEM and DBPM were used for the prediction of network and hardware features, and the communication frequency and SVC multimedia streaming files most suitable for the device environment were determined according to these two modules. In the experiment, the overall prototype architecture was realized and an experimental analysis was carried out. The experimental data proved that the method could maintain a certain level of multimedia service quality for dynamic network environments and ensure smooth and complete multimedia streaming services. Cloud services may accelerate research on SVC coding in the future. This study presented a network and device-aware Quality of Service (QoS) approach that provides multimedia data suitable for a terminal unit environment via interactive mobile streaming services, further considering the overall network environment and adjusting the interactive transmission frequency and the dynamic multimedia transcoding, to avoid the waste of bandwidth and terminal power. Finally, this study realized a prototype of this architecture to validate the feasibility of the proposed method.

REFERENCES


