Bandwidth Estimation and Optimized Bitrate Selection for Dynamic Adaptive Streaming Over HTTP Using RSI-GM and ISSO

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ABSTRACT

Dynamic adaptive streaming over HTTP (DASH) is an emerging solution that aims to standardize existing proprietary streaming systems. DASH specification defines the media presentation description (MPD), which describes a list of available content, URL addresses, and the segment format. High bandwidth demands in interactive streaming applications pose challenges in efficiently utilizing the available bandwidth. In this paper, a novel relative strength index (RSI) with geometric mean (GM), namely RSI-GM, is proposed for estimating available bandwidth for DASH. The work starts by taking the video as an input at the transmitter side and then the video compression is performed using the TRLE. Then MD5 hashing-based AES encryption is applied to the compressed video data to provide data security. Then RSI-GM is proposed to estimate the available bandwidth for DASH. Finally, after estimation, the bitrate for estimated bandwidth is selected optimally using the improved shark smell optimization (ISSO) algorithm.

KEYWORDS

Advanced Encryption Standard, Bandwidth Estimation, Bitrate Selection, Dynamic Adaptive Video Streaming Over HTTP (DASH), Shark Smell Optimization (ISSO), Video Compression, Video Encryption

INTRODUCTION

The significance and usage of multimedia traffic are rapidly increasing over the last few years Hassan et al. (2020). As video traffic has grown, many commercial video providers have employed adaptive bitrates streaming techniques to provide streaming media with the best experience for users Kim et al. (2017). HTTP adaptive streaming technologies include MPEG-DASH, Apple’s HTTP Live Streaming, Microsoft’s Smooth Streaming, Adobe’s Dynamic Streaming, etc Wu et al. (2017). Due to the heterogeneity of today’s communication networks, adaptively is one of the most important requirements for any streaming systems Le et al. (2018). Accurate bandwidth estimation is an
important task as it regulates the user’s buffer and influences the user-perceived Quality of Service (QoS) Mushtaq and Mellouk (2017). However, there are many challenges for providing satisfactory levels of quality-of-service (QoS) to end-users, such as bandwidth-constrained, variable capacity links, and energy-constrained operation Castellanos et al. (2019).

HTTP Adaptive Streaming (HAS) can adjust video quality to the most appropriate level on a moment-by-moment basis according to the current network condition, for example, the available network bandwidth Hwang et al. (2016). In addition, this technology can bring a chance to view videos for different users with different Internet connection capacities Phan-Xuan and Kamioka (2016). To provide the user the seamless multimedia service with maximum achievable Quality of Experience (QoE), the media content in a particular video needs to be adaptive to match the available bit rate in the network Kumar et al. (2016). This trend is expected to continue as the Internet infrastructure has evolved to support HTTP, and HTTP is firewall-friendly ur Rahman and Chung (2017).

Recently, commercial streaming services, such as Netflix and YouTube, have begun employing HAS as their default delivery method Yun et al. (2018). The 2015 Conviva report shows that almost 30% of the analyzed HAS sessions are affected by at least one freeze Petrangeli et al. (2017). Initially, Internet video streaming was implemented by using traditional streaming protocols, such as Real-Time Protocol (RTP) over User Datagram Protocol (UDP). But since firewalls usually block UDP packets in the networks, making it hard to deliver the content to the user Ayad et al. (2018).

DASH (dynamic adaptive streaming over HTTP) is a stream-switching adaptive protocol that is based on the download of portions of a video called segments or chunks over HTTP-connections Batalla (2016). The Quality of Experience (QoE) for a viewer in adaptive video streaming can be enhanced by minimizing frequent playback interruptions Kumar et al. (2015). With DASH, a video is segmented into short segments encoded at various bit rates. This information is then stored in a media presentation description (MPD) file Tashtarian et al. (2018). The bandwidth-based method is to select the video bitrate as close to the actual bandwidth as possible Du et al. (2018). Thus, DASH is able to provide the best possible video quality to the clients. However, due to the inherent bandwidth fluctuation, challenges to estimate accurately current bandwidth still exist, especially when multiple DASH clients share the same bottleneck link Van-Huy et al. (2016). In the actual streaming strategy, the client intelligence for selecting the right quality for each chunk in order to produce a high quality of experience (QoE) for the viewer is left to the developers Gong et al. (2015). Various kinds of buffer and throughput-based methods for adaptive video streaming are surveyed in section 2. However, they are suffering from the following drawbacks: i) decreasing the QoE in the case of long-term bandwidth variations and slow bandwidth fluctuation detection, and ii) video streaming is not optimized from the QoE point of view. In addition, traditional Adaptive Bitrate Streaming (ABR) algorithms mainly focus on improving Internet video streaming quality where network conditions are relatively stable. These approaches, however, suffer from performance degradation at any of the networking platforms like the Internet of Things (IoT) edge. In IoT systems, the wireless channels are prone to interference and malicious attacks, which significantly impact QoE for video streaming applications. To tackle these issues, in this paper, a novel RSI-GM is proposed for performing efficient bandwidth estimation in DASH. Additionally, the paper proposes an ISSO-based optimal bitrate selection for video streaming. The objectives of the current study are enlisted as follows:

- To present a novel Transform based Run-length Encoding (TRLE) methodology for efficient video compression.
- To present an efficient encryption scheme namely MD5-AES128 for providing security to the video contents.
- To present a novel RSI indicator with two sensitive thresholds and a geometric mean (GM) to estimate the available bandwidth.
- To present an ISSO to determine the next multimedia bitrate for video streaming.
The draft structure of this paper is systematized as Section 2 surveys the associated works regarding the proposed method. A brief elucidation of the proposed work is proffered in section 3. Section 4, explores the experimental outcome, and the conclusion is exhibited in section 5.

RELATED WORK

Jingyan Jiang et al. Jiang et al. (2018) developed an online Q-learning-based dynamic bandwidth allocation algorithm Q-FDBA with the goal of QoE fairness. In this method, a pure network-based architecture used Software Defined Network was designed for monitoring network global performance information. This method was designed and implemented in mininet-based testbed. The results showed that the Q-FDBA could adaptively react to the high frequency of bottleneck bandwidth switches and achieve better QoE fairness within a certain time dimension. The playback freeze could be reduced and the bitrates switch could be smoother than baseline algorithms. However, this technique needs more time for bandwidth allocation.

Ran Dubin et al. Dubin et al. (2019) presented the Fair Server Adaptation (FSA) algorithm, which was designed to maximize user Quality of Experience (QoE) by tackling the server’s bottleneck problem. The algorithm provided the quality representation that was closest to the user’s request, subject to the server’s constraints. Simulation results show that compared to standard Dynamic Adaptive Streaming over HTTP (DASH) server, FSA increased the number of served users and decreased both the number of rebuffering events and the average rebuffering event duration. It was clear that these changes increased users’ QoE. This method does not consider the mobile link limitations and mobile base station.

Nabin Kumar Karn et al. Karn et al. (2019) developed the receiver-side bandwidth-measured method to achieve a better quality of experience in the multiple-client scenario. This method estimated the obtainable network bandwidth based on the buffer status and segment throughput. The video buffer model was associated with three thresholds (i.e., one for the initial start-up and two for operating thresholds). NS-3 network simulator had been deployed to measure the performance of HTTP adaptive streaming. Simulation outputs reflected that this method enhanced the quality of experience than conventional methods. However, this causes the underutilization problem because the bandwidth offered by the server was quantized and limited.

Pham Hong Thinh et al. Thinh et al. (2020) presented an architecture that incorporated bitrate adaptation and dynamic route allocation. On the client side, the adaptation logic of VBR videos streaming was built based on the MPEG-DASH standard. On the network side, a Software-Defined Networking (SDN) controller was implemented with several routing strategies on top of the OpenFlow protocol. Results showed that the solution enhanced at least 38% up to 185% in terms of average bitrate in comparison with some existing solutions as well as achieved a better viewing experience than the traditional Internet. However, the problem was that the topology was more complicated and multiple clients connect to multiple servers.

Zeng zeng et al. Zeng et al. (2020) presented a secure and network-state-aware solution, SASA for performing bitrate adaptation at the Internet of Things (IoT) edge. Firstly, the buffer-level constraint was studied when the bitrate was increased. Then, the impact of throughput overestimation in bitrate decisions was analyzed. Based on these results, SASA was designed to consist of both an offline and an online phase. In the offline phase, SASA pre-computed the best configurations of ABR algorithms under various network conditions. In the online phase, SASA adopted an online Bayesian change point detection method to detect network changes and applied pre-computed configurations to make bitrate decisions. SASA was implemented and its performance was evaluated using 429 real network traces. The SASA outperformed state-of-the-art ABR algorithms, such as RobustMPC and Oboe in the IoT environment through extensive experiments. However, this method was sensitive to highly variable throughput, such as short-term bitrate peaks. In these situations, the client will request a high-quality level and cannot be able to completely download a segment before the throughput suddenly falls.
Luca De Cicco et al. Cicco et al. (2019) presented ERUDITE, a closed-loop system to optimally tune at run-time the adaptive streaming controller’s parameters to adapt to changing scenario’s parameters. The system was based on a Deep Neural Network (DNN), which continuously provided the streaming controller with estimates of optimal parameters based on measured metrics, such as bandwidth samples and overall obtained QoE. The DNN was trained using a dataset that was built by finding, for thousands of scenarios, the optimal adaptive streaming controller’s parameters using a Bayesian optimization algorithm. Results, gathered considering a large number of diverse scenarios, showed that ERUDITE was able to provide near-optimal performances by reducing impairments due to re-buffering and video level switching. However, the results of Bayesian optimization were sensitive to parameters of the surrogate model, which were typically fixed at some value; this underestimated the uncertainty.

PROPOSED METHODOLOGY

Dynamic adaptive video streaming over HTTP (DASH) has been developed as one of the most suitable technologies for the transmission of live and on-demand audio and video content over any IP network. DASH allows the client to make adaptation decisions based on available bandwidth. To offer a better user experience and cope with dynamically varying network conditions, it is necessary to estimate precisely the next segment bandwidth, which is a significant premise in determining the next multimedia bitrate, especially when multiple clients share the same bottleneck link. In this paper, a novel RSI indicator is presented with two sensitive thresholds and a geometric mean to estimate the available bandwidth. Initially, at the transmitter side, the video is taken as an input and they are converted into multiple video frames. The converted video frames are compressed in order to reduce the size and to increase the speed of the system using a novel Transform based Run-length Encoding (TRLE). Next, the compressed data is encrypted using the MD5-AES128 algorithm for securely transmitting the video data. Then an innovative mechanism (RSI-GM) is used to estimate the bandwidth based on the RSI indicator and its two thresholds. Finally, with the estimated bandwidth, ISSO is proposed to determine the next multimedia bitrate for video streaming. On the receiver side, the inverse operations are performed to get original input data. The architecture of the proposed method is shown in figure 1.

Video Compression

The first phase of the proposed method is video compression. Video compression is performed because it allows real-time video streams or the resultant files to be easily transmitted across today’s standard networks. It is also very beneficial for the efficient use of storage due to the dramatically reduced file sizes. For compressing the video file, the input video of the transmitter is firstly converted into multiple frames. Then, on the converted frames, the proposed TRLE is applied to get (encoded) compressed frames. Run-length encoding (RLE) is a form of lossless data compression in which runs of data (sequences in which the same data value occurs in many consecutive data elements) are stored as a single data value and count, rather than as the original run. This is most useful on data that contains many such runs.

Apply DWT on the converted frames of the video, and then select only important coefficients and apply RLE for encoding. The discrete wavelet transform (DWT) is a computerized technique to compute the fast wavelet transform of an image. The DWT is an optimum solution for computational time overhead. In DWT operation, an image can be analyzed by the combination of analysis filter bank and decimation operation. The analysis filter bank consists of a pair of low and high pass filters corresponding to each decomposition level. The low pass filter extracts the approximate information of the image whereas the high pass filter extracts the details, such as edges. The proposed method uses 2D DWT for extraction coefficients from the converted video frames. The 2D DWT decomposes the input video frame into four separate sub-bands: low-frequency components in horizontal and
vertical directions (after 1-level DWT with its sub-bands is expressed) where (smaller scaled form) and sub-band successively and the resultant image is split into multiple bands. A video frame after 2-level DWT decomposition is represented.

The use of DWT allows the transformed data to be sorted at a resolution, which matches its scale. Then from the DWT coefficients, only significant coefficients are selected for encoding. The number of coefficients depends on the quality and compression ratio (CR) to be obtained, discrete wavelet coefficients with larger magnitudes are considered more significant than coefficients with smaller magnitudes. The most significant coefficients are kept while dropping all the others. A large number of significant coefficients lead to high PSNR. This procedure leads to a sequence of the zero-valued number inside the elements of the directional sub-bands, a fact exploited by using RLE in order to achieve higher CR. This transform domain-based RLE is called TRLE for video compression.

Data Encryption

After video compression, to provide security to the transmitter side data, the compressed data is encrypted and shared. The paper uses the MD5-AES128 bit algorithm for performing encryption on the compressed data. The more popular and widely adopted symmetric encryption algorithm likely to be encountered nowadays is the Advanced Encryption Standard (AES). It is based on ‘substitution-permutation network’. It comprises a series of linked operations, some of which involve replacing inputs with specific outputs (substitutions) and others involve shuffling bits around (permutations). AES includes three block ciphers: AES-128, AES-192, and AES-256. AES-128 uses a 128-bit key length to encrypt and decrypt a block of messages, while AES-192 uses a 192-bit key length and AES-256 a 256-bit key length to encrypt and decrypt messages. Each cipher encrypts and decrypts data in blocks of 128 bits using cryptographic keys of 128, 192, and 256 bits, respectively. The proposed method uses the 128 bit key of AES with improved features for performing encryption on the video data (compressed video frames). The improved features include the key generation of AES with the

Figure 1. Proposed architecture
combination of salt value and MD5 hash function. The objective of using the MD5 hash function is to make the key value of the AES complex. The new key is generated by concatenating salt value.

It proves that after including salt value and hash function in AES, the AES became improved AES and it is named MD5-AES128 encryption. After key generation, the compressed video frames are encrypted using the MD5-AES128 algorithm. The workflow of MD5-AES128 is shown in figure 2.

The number of rounds in the proposed algorithm is variable and depends on the length of the key. Generally, AES uses 10 rounds for 128-bit keys, 12 rounds for 192-bit keys, and 14 rounds for 256-bit keys. The proposed method uses the 128 bit key for performing both encryption and decryption. Each round comprises four sub-processes: Sub Bytes, Shift Rows, Mix Columns, and Add Key to encrypt the data. Before performing the rounds, the key expansion is performed. In this, the generated 128-bit key where:

1. **Byte Substitution**: The 16 input bytes are substituted by looking up a fixed table (S-box). The result is in a matrix of four rows and four columns.
2. **Shift rows**: Each of the four rows of the matrix is shifted to the left. Any entries that ‘fall off’ are re-inserted on the right side of the row. The shift is carried out as follows:
   a. First row is not shifted.
   b. Second row is shifted one (byte) position to the left.
   c. Third row is shifted two positions to the left.
   d. Fourth row is shifted three positions to the left. The result is a new matrix consisting of the same 16 bytes but shifted with respect to each other.

Figure 2. Workflow of MD5-AES128
3. **Mix Columns**: Each column of four bytes is now transformed using a special mathematical function. This function takes as input the four bytes of one column and outputs four completely new bytes, which replace the original column. The result is another new matrix consisting of 16 new bytes. It should be noted that this step is not performed in the last round.

4. **Add round key**: The 16 bytes of the matrix are now considered as 128 bits and are XORed to the 128 bits of the round key. If this is the last round, then the output is the ciphertext. Otherwise, the resulting 128 bits are interpreted as 16 bytes, and the algorithm starts another similar round.

**Bandwidth Estimation**

In DASH, a client dynamically selects an appropriate bitrate of the media segment in the MPD file based on the network condition. For this, it is necessary to address two principal issues: current bandwidth estimation and determination of suitable bitrate for the estimated bandwidth. This paper uses the method, which is based on the RSI indicator and its movement to determine the degree of change in the available bandwidth, small or large. To smooth the varying estimated bandwidth and avoid the transient change of available bandwidth leading to the unnecessary oscillation of estimated bandwidth, geometric mean (GM) is used to deal with this purpose. The advantages of using GM are: fluctuation in sampling will not affect the GM, and it gives relatively more weight to small observations. The encrypted video frames are converted into video segments, and then, the bandwidth is estimated by applying the RSI indicator. Two thresholds are set to determine small or large fluctuations of the network bandwidth. Then the GM is used to smooth the estimated bandwidth. This combination of GM and RSI for bandwidth estimation is named RSI-GM. The relative strength index (RSI) is a technical indicator used in the analysis of financial markets. It is intended to chart the current and historical strength or weakness of a stock or market based on the closing prices of a recent trading period.

When the RSI indicator is applied to detect the fluctuation momentum of estimated bandwidth, the system not only can adapt to the network condition change in a long period but also ignore the drastic fluctuation in a short period. RSI values are plotted on a scale from 0 to 100; if the RSI is below 70, bandwidth tends to decrease or the network bottleneck may have occurred, and vice versa. If it is above 70, bandwidth tends to increase or network bandwidth may have increased. Because any change of estimated bandwidth directly affects multimedia bitrate, so two thresholds namely and are set to above or below 70 to classify if the fluctuation is small or large, which is mathematically expressed.

**Optimal Bitrate Selection**

In DASH, the bitrate of a video is not constant. Clients can dynamically select an appropriate bitrate based on its network condition from the list of media bitrates which is provided by MPD. There are some issues in bitrate selection. Firstly, if the instant video bitrate selected by the client is much less than the available bandwidth, the service quality is low while the network bandwidth is poorly used. Secondly, if the instant video bitrate is greater than the available bandwidth, the result is buffer underflow and frequent interruption during video playback. Thus, choosing an appropriate bitrate for a video is important. When a number of competing clients share the same link, selecting media bitrate plays an important role not only to guarantee the service quality but also to optimize bandwidth utilization.

To achieve optimal bitrate selection and user perceive quality, this work uses the ISSO algorithm to solve the problem. ISSO is proposed to determine the bitrate in each channel under the following assumptions: The input of ISSO consists of the total estimated bandwidth, number of channels, number of clients per channelin each channel provided by MPD files. The output of ISSO includes an optimal list of bitrates for each channel. Generally, the Shark smell optimization algorithm (SSO) algorithm is a metaheuristic method inspired by the hunting capability of the shark by the sense of smell of the prey odor. SSO process is based on the following considerations:
- The prey is wounded and released blood into the seawater. Hence, the speed of the victim is lower than shark speed and even negligible. Though, the prey is considered to be stable and fixed.
- The blood is continuously released into the water, and the impact of the seawater flow on contorting the prey odor particles is neglected.
- There is one source of blood injection (one prey); there is only one seeking environment.

However, the basic SSO algorithm has the disadvantage of search stagnation, easily falls into a local optimum, has slow convergence speed, and has low calculation accuracy. To overcome the premature convergence problem of the SSO algorithm, this paper proposes an Improved SSO (ISSO) with the integration of the adaptive inertia coefficient and levy flight mechanism. The search process begins when the shark smells odor. In fact, the particles of odor have a weak diffusion from an injured fish (prey). To model this process, a population of initial solutions is randomly generated for an optimization problem in the feasible search space. Each of these solutions represents a particle of odor which shows a possible position of the shark at the beginning of the search process.

The movement types, as mentioned above, are iteratively repeated until a termination criterion is reached. The pseudo-code of the proposed ISSO is illustrated in figure 3. The above phases are done at the transmitter side, and for each block on the receiver side, inverse operations are performed. The data are decrypted using the MD5-AES128 algorithm, then decrypted data is compressed using the WTRLE, and finally, the receiver obtained the input video.

Figure 3. Pseudo-code of the ISSO
RESULTS AND DISCUSSION

In this paper, two methods namely RSI-GM and ISSO are proposed to efficiently estimate the available bandwidth and to select the optimal bitrate for DASH. The proposed work is implemented in the working platform of MATLAB. In this section, the results of proposed techniques such as video compression, encryption, and bandwidth estimation are compared with the existing techniques regarding some performance metrics. Each of the technique with their comparative analysis is explained in the below sections.

Performance Analysis of Video Compression

The performance of the proposed TRLE is analyzed with regards to the compression time and compression ratio. The Proposed TRLE algorithm is compared with the existing methods such as Unicode based Huffman Encoding (UHE), High Run-length encoding (RLE), Lempel Ziv Welch, and fractal encoding. The diagrammatic representation of the performance metrics are explained in Figures 4 and 5.

Figure 4 exhibits the Proposed TRLE’s performance with that of the conventional UHE, RLE, Lempel Ziv Welch, and fractal encoding centred upon the (a) compression time (b) compression ratio. Based on the analysis of compression time metric, the fractal encoding, Lempel ziv Welch, RLE and UHE takes 51,890ms, 33,768, 27,856ms, 19,854ms time to compress the video, whereas the Proposed TRLE method takes 16,524ms time to compress the video, which takes lesser than the existent methodologies. Concerning the compression ratiometric, the Proposed TRLE achieves a 91.568 of compression ratio whereas, the UHE, RLE, Lempel Ziv Welch, and fractal encoding have a compression ratio of 81.568, 70.189, 56.953, and 49.667 respectively. Here, the fractal encoding gives a very low value of compression ratio and the UHE technique works well than fractal encoding. But compared to all, the proposed one attains the highest level of compression ratio.

Figure 4. Comparative analysis of the suggested method with the existent method in terms of compression time
Performance Analysis of Secure Data Transfer

Next, the performance of the proposed MD5-AES128 is contrasted with the traditional methods such as Koblitz’s Encoding based Elliptic Curve Cryptography (KEECC), Rivest–Shamir–Adleman (RSA), Triple Data Encryption Standard (Triple DES), Blowfish with respect to some performance metrics, namely, encryption time and decryption time. The compared descriptions of the encryption time and decryption time is explained in Tables 1 and 2.

Tables 1 and 2 depict the performance of the proposed MD5-AES128 algorithm with the existent KECC, RSA, Triple DES and Blowfish methods in terms of encryption time and decryption time. The time taken by diverse techniques to execute encryption along with decryption for a disparate number of data sizes (5 to 25) is contrasted. From Table 1, for data size 5 Mb, the proposed MD5-AES128 algorithm takes 489ms time to encrypt the data, whereas the existent KECC, RSA, Triple DES and Blowfish algorithm take 512ms, 936ms, 1025ms, and 1225ms time to encrypt the data, which is higher than the proposed methodology. From Table 2, for data size 5 Mb, the existent KECC, RSA, Triple

| Size  | Proposed MD5-AES128 | KEECC | RSA | Triple DES | Blowfish |
|-------|---------------------|-------|-----|------------|----------|
| 5(Mb) | 489                 | 512   | 936 | 1025       | 1225     |
| 10    | 1005                | 1112  | 1646| 1826       | 2001     |
| 15    | 2068                | 2157  | 2778| 2873       | 2999     |
| 20    | 3015                | 2589  | 3979| 4041       | 4142     |
| 25    | 3999                | 4021  | 5008| 5113       | 5524     |
DES and Blowfish takes 459ms, 776ms, 874ms and 1007 ms, whereas the proposed MD5-AES128 algorithm takes 405ms time to decrypt the data. Likewise, the proposed methodology attains higher performances for the remaining data sizes from 10kb to 25kb. Therefore, the discussion shows that the proposed method achieves higher performance than the existent methodologies.

### Performance Analysis of Bandwidth Estimation

The performances of the proposed bandwidth estimation using RSI-GM are compared with the existent Weight and Levy flight based Black Widow Optimization (WLBWO), particle swarm optimization (PSO), Butterfly Optimization Algorithm (BOA), Shark Smell Optimization (SSO). The performance comparison is done by using some qualitative performance metrics, namely, signal-to-noise ratio (SNR) and Mean Squared Error (MSE).

Figure 6 illustrates the performance of the proposed RSI-GM algorithm with the conventional WLBWO, SSO, PSO and BOA algorithm in terms of MSE based on the SNR performance metrics. The error for the techniques is identified based on the SNR value. Here, the error for the techniques

| Size (Mb) | Proposed MD5-AES128 | KEECC | RSA | Triple DES | Blowfish |
|----------|----------------------|-------|-----|------------|----------|
| 10       | 997                  | 1023  | 1478   | 1661       | 1772     |
| 15       | 2116                 | 2378  | 3099   | 3202       | 3927     |
| 20       | 3356                 | 3987  | 3378   | 3665       | 4015     |

The error for the techniques is identified based on the error value. Here, the error for 

![Figure 6. Comparative analysis of the bandwidth estimation with the existing methodologies](image.png)
is estimated for 10 numbers of SNR. For SNR, the proposed one gives the error of 0.21 whereas the existing WLBWO, SSO, PSO and BOA give the error of 0.26, 0.69, 0.53, and 0.67, which is higher than RSI-GM. The error of the existing method is increased when the number of SNR increases but the proposed RSI-GM gives the lowest errors for all 10 SNR when compared to existent methodologies. So it is concluded that the proposed method minimizes the error contrasted to conventional methodologies.

Figure 7 show the comparison of the estimated bandwidth with various cross traffics. Proposed RSI-GM shows stable performance in the cross-traffic environment. Also, the bandwidth measured by wbest has a huge variance in every cross traffic, which means that it cannot give accurate information. EBE significantly underestimated the bandwidth. The proposed RSI-GM has a high value of available bandwidth compared to all existing methods.

Figure 8 shows the comparison of the convergence time. The value of pathlaod indicates a slow measurement time (6.5s). Wbest sets the measurement time of 7.5 seconds. Similarly, the Pathchirp and Efficient Bandwidth Estimation (EBE) sets 5.2s and 6.2s. The proposed RSI-GM scheme shows the 1.2 seconds measurement time. Thus, the proposed RSI-GM indicates the speedy measurement time compared to all existence methodologies.

CONCLUSION

This paper proposes a novel RSI indicator with two thresholds and GM - (RSI-GM) for estimating the available bandwidth for DASH. Additionally, a bitrate for adaptive video streaming (DASH) is selected optimally using the improved version of SSO algorithm. The results of the proposed techniques used in compression, encryption, and bandwidth estimation are compared with the previous algorithms to analyze its performance efficiency. The proposed compression algorithm (TRLE) obtains the best results for both CR and CT. it attains highest Cr and lowest CT than the existing algorithms. Likewise, when analyzing the results of proposed MD5-AES128 bit algorithm for data security, it attains the...
lowest time for both encryption and decryption process when compared with Triple DES, KEECC, RSA, and Blowfish. Finally, when comparing the results of proposed RSI-GM with existing algorithms, the MSE and convergence time obtained by the RSI-GM is very low. These results show that the proposed method of bandwidth estimation using RSI-GM works well when compared to previous algorithms. Furthermore, the proposed bandwidth estimation method is also sensitive to multiple levels of bandwidth changing. Besides, the method has a smooth value in estimating bandwidth in both stable state and agile state. In the future, the work will be extended by applying multiple levels of thresholds for RSI indicator in bandwidth estimation.

Figure 8. Convergence time of the techniques
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