



# QoS-guaranteed Mobile IPTV service in heterogeneous access networks



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## ABSTRACT

Internet Protocol Television is defined as a multimedia service delivered over IP-based networks supporting quality of service (QoS), quality of experience (QoE), security, interactivity, and reliability. This service is rapidly expanding to both wireless and mobile networks through mobile devices. This trend demands a seamless IPTV service architecture for those mobile devices in heterogeneous access networks. This is because the convergence of the heterogeneous access networks can resolve their own service-coverage limitations, eliminating dead spots. We first briefly research on the background and trend for the seamless IPTV service for heterogeneous networks. Next we propose a new system architecture taking the relevant technical issues into account. This architecture is mainly composed of three parts, (i) information of network conditions on client, (ii) signaling for communicating between client and server with information of network conditions, and (iii) adaptive streaming based on information of network conditions. The proposed architecture can be deployed on any Internet protocol layers such as application layer, transport layer and network layer. In this paper, we evaluate two cases such as (i) RTSP in application layer and (ii) SCTP in transport layer and compare them through the performance evaluation. The results in the evaluation show that the proposed architecture meets well the requirements of the minimum IPTV service performance recommended by ITU-T international standard for the QoE over heterogeneous access networks.

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## 1. Introduction

Internet Protocol Television (IPTV) service has rapidly been expanded to wireless mobile areas. In other words, users are able to use IPTV services everywhere and even in motion. The traditional IPTV service targets for the quality-of-service (QoS) and the quality-of-experience (QoE) guaranteed wired networks which should provide at least 10 Mbps according to the ITU-T specification. Therefore,

seamless IPTV service in wireless mobile environments (a.k.a. Mobile IPTV) [1] must overcome several technical obstacles for the commercial service. Mobile IPTV implies at least one wireless network between the source and the destination. Therefore, most of the technical challenges are related to the lack of bandwidth and the mobility of mobile devices in the wireless networks.

In particular, mobile devices equipped with multiple access technologies including wired and wireless network interfaces are becoming common. Consequently, more frequent handovers between different access technologies become required. These mobile devices (e.g., mobile phones, smartphones or even laptops) may be reachable through

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multiple interfaces even simultaneously. The possibility of using a single or multiple interfaces at a time for sending and receiving IP packets depends on the mobile device capabilities. In both cases, handover between heterogeneous networks (a.k.a. vertical handovers) can occur. A significant change in the access network as a result of a handover may also affect End-to-End (E2E) path properties such as bandwidth, latency, data throughput, bit-error rate, and so on. Eventually, this situation makes IPTV systems have more challenges to provide seamless IPTV services with IPTV mobile devices equipped with multiple network interfaces.

The rest of this paper is organized as follows. Section 2 shows the background and motivation of Mobile IPTV in brief. Section 3 describes related work and Section 4 describes our Mobile IPTV system architecture, service modeling, and vertical handover decision algorithm. Section 5 describes evaluation criteria based on the ITU-T requirements. Based on the system architecture and evaluation criteria, we describe two approaches such as RTSP-based approach in Section 6 and SCTP-based approach in Section 7, respectively. Finally, Section 8 concludes this paper along with future work.

## 2. Background and motivation

IPTV service expansion has several technical issues when in heterogeneous access networks with the support of handover, the IPTV may affect E2E path properties such as bandwidth, latency, data throughput, and bit-error rate [2]. Note that these properties are parts of Information of Network Characteristics (INC). The changes of the INC in the local access network and consequently those in the E2E path are usually not detected nor reacted quickly enough by higher layer transport protocols and applications. Typically the higher layers do not react to the changes in path properties until certain mechanisms, such as congestion control or error recovery, eventually get invoked at some point later. This may cause undesirable disruptions, performance degradation to the ongoing connections, unnecessary underutilization of the available network capacity, or sudden overloading of the next access network involved in handover. This can lead to the dissatisfaction for IPTV service in the expansion to more heterogeneous access networks.

For example, in the case of Mobile IP [3] where a mobile node performs a handover from IEEE 802.11b WLAN network (high bandwidth link) to CDMA Cellular network (low bandwidth link), the home agent and correspondent nodes may still continue transmitting at the rate adapted to 802.11b bandwidth. As the actual path capacity becomes now smaller, a packet loss burst will occur and often result in inefficient loss recovery at the transport protocol level. This situation could be resolved by explicitly informing the other connection peer of the significant changes of the INC in the local access network. Unfortunately, existing IP mobility, transport and application layer protocols do not provide any facility to indicate which type of network the mobile node is currently attached to or what kind of changes happened on the local access network.

As of now, a wired network is used for the communication between the fixed IPTV device at home (via set-up

box as usual) and the IPTV streaming server at IPTV service provider. Recently, a wireless network is popularly used for the communication between the Mobile IPTV device and the IPTV streaming server in both indoor and outdoor via various wireless network technologies such as Wireless Local Area Network (WLAN, e.g., Wi-Fi) and Wireless Wide Area Network (WWAN, e.g., WiMAX and Cellular Networks such as 3G, 4G and even LTE-Advanced) as depicted in Fig. 1. The conditions of the local INC may also vary significantly as a result of a handover between the networks of the same type (called horizontal handovers).

For example, the current network may have significantly more traffic load than the previous network or the new route taken by the IP traffic may have different E2E path properties. Moreover, even if the mobile node stays on the same network, the conditions of the local INC may change significantly due to various reasons, for example, because of sudden variations of the traffic load on the current network. All of these situations may lead to similar as much negative effects as those with vertical handovers. Therefore, to support the seamless IPTV service in heterogeneous access networks, a Mobile IPTV system architecture becomes required at this point. In next section, for this seamless IPTV, we will propose a Mobile IPTV system architecture.

## 3. Related work

In this section, we introduce related work in terms of seamless streaming multimedia services over the Internet. Nowadays, streaming contents over the Internet are provided by multiple transport protocols for data delivery rather than the unified solution. Most of significant example is Real Time Streaming Protocol (RTSP) [4] that is making the transmission of data even more efficient than other previous protocols. Especially, RTSP is ideal for video broadcasting since it places a high priority on continuous streaming rather than on other factors. However, the biggest issue with RTSP is that the protocol or its ports may be blocked by routers or firewall settings, preventing a device from accessing the streaming contents.

In contrast to RTSP, Hypertext Transfer Protocol (HTTP) [5] is more widely supported in content distribution networks and does not depend on any special sever rather than a standard HTTP server. This is why HTTP is more popularly used than RTSP. Also HTTP is generally accessible and allowed to traverse firewalls using TCP port 80, which can facilitate the HTTP delivery of contents in many cases. As the standard protocol for the Web, HTTP is originally designed to reliably transfer data (e.g., text documents, email, executable programs, and HTML web pages) over the Internet, while enforcing maximum reliability and data integrity rather than timeliness. However, when HTTP is used to transmit the streaming contents relying on time-based operation, it is much more likely to cause major packet drop-outs due to TCP based retransmission for packet loss, and it cannot deliver nearly the same amount of streams as RTSP transmission. Along with RTSP, various protocols can be used for supporting seamless IPTV service in heterogeneous access networks. Regarding contents streaming in Mobile IPTV, an IPTV architecture should

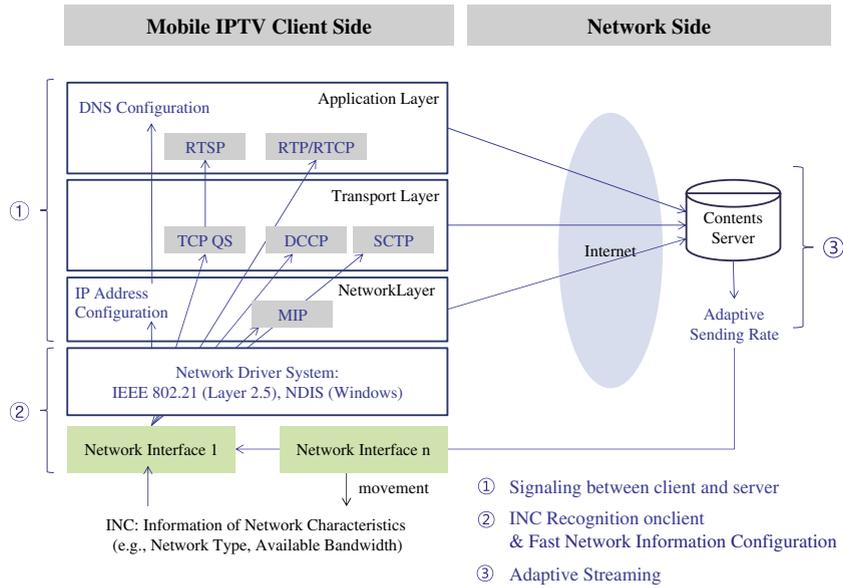


Fig. 1. Mobile IPTV overall architecture.

not be biased to any specific protocols such as RTSP and HTTP, and allow for IPTV service providers to adopt their own preferable solutions, based on their requirements and considerations.

As an example of the transport layer approach, SCTP [6] was developed by the IETF in 2000 and used for call control signaling in voice over IP networks; since SCTP originally was intended to be used for the transport of telephony over Internet protocol. SCTP is characterized as message-oriented, meaning it transports a sequence of messages rather than transporting an unbroken stream of bytes as in TCP. As in UDP, a sender in SCTP transmits a message toward its message destination in one operation and then the exact message is passed to the receiving application process (i.e., the message destination) in one operation. The related work [7] suggested SCTP as the transport protocol in IPTV system. In particular, Partial-Reliable SCTP (PR-SCTP) [8] was used to provide timely reliability services for transporting real-time IPTV traffic. It is noted in PR-SCTP that the control or adjustment of lifetime may make a significant impact on the throughput performance of SCTP. To decide the precise lifetime value, there are several considerable factors such as network transmission delay, packet loss and the upper layer applications requirements. The general rule to determine the lifetime for each application message is described as follows:

$$Lifetime_N = R_{APP} + \frac{L_N}{B_{MEDIA}} \times 1024, \quad (1)$$

where  $Lifetime_N$  is the lifetime [ms] of application message  $N$ ,  $R_{APP}$  is the delay [ms] according to rate control,  $L_N$  is the length [bytes] of application message, and  $B_{MEDIA}$  is the average bit rate [bps] of media. In (1), it is assumed that the upper layer application adjusts the value of  $R_{APP}$  according to the state of application buffer, network delay and the related network characteristics. If no rate control is

allowed, the value of  $R_{APP}$  will be set zero. By using the equation above, real-time IPTV traffic can be delivered within the specific time limit. There is a pitfall for PR-SCTP to provide seamless IPTV service in heterogeneous access networks due to lack of function for deciding the available network bandwidth. Unless otherwise the available network bandwidth is supported for PR-SCTP, lifetime cannot be set correctly. It is difficult for the PR-SCTP sender to determine a reasonable requested rate since it usually has no good knowledge about the network conditions. So, the proposed architecture in this section does not adopt PR-SCTP for seamless IPTV service in heterogeneous access networks. Instead, we simply designed the system with SCTP as to piggyback INC extracted in the IPTV mobile device to the SCTP sender.

#### 4. Mobile IPTV system architecture

In this section, first, we will explain the overall Mobile IPTV system architecture. Second, we will formulate the signaling for the INC delivery in heterogeneous access networks. Also, we will explain the vertical handover decision algorithm when the vertical handover occurs in the proposed networks including Wi-Fi, WiMAX and 3G. The vertical handover is essential for seamless service in the architecture of the forthcoming heterogeneous networks. To offer systematic comparisons, we research on the related algorithms and select an appropriate decision algorithm based on the cost function. The cost factors are extracted from the IPTV QoE requirements [2] such as bandwidth and stabilization time. Regarding the adaptive streaming, we will address the related approaches and algorithms, and come up with the acceptable solution for the proposed architecture.

Mobile IPTV architecture is required to allow the delivery of IPTV services to any IPTV devices over different

access networks including wired and wireless. Also, Mobile IPTV architecture needs to allow service continuity over heterogeneous networks through handover. Regarding signaling between client and server, Mobile IPTV architecture is required to support signaling capabilities for transmitting bandwidth-related information. In addition, it needs to support capabilities for the interoperability and user mobility between IPTV networks, allowing customer access to IPTV services whether or not the customer is mobile. Based on signaling, Mobile IPTV architecture is required to adapt dynamically to change in wireless networks characteristics, such as bandwidth and packet-loss ratio, when the system delivers the service over mobile networks. Once getting the network characteristics, Mobile IPTV architecture needs to deliver contents in several optional versions to be selected according to the capabilities of the Mobile IPTV client receiving the content, and support the Mobile IPTV client with the capability to choose the desired contents format if multiple formats are available. To satisfy these requirements for Mobile IPTV services, we propose Mobile IPTV architecture with the interaction between Mobile IPTV Client and Contents Server in the Internet, as depicted in Fig. 1. Note that the IP address configuration and DNS configuration as depicted in Fig. 1 are not illustrated in this paper and more information can be found in [9,10].

Therefore, we try to design and propose a general, flexible Mobile IPTV architecture for upcoming the new IPTV service in near future. The proposed architecture should be compatible and acceptable for the existing IPTV service and network model. For this purpose, we try to adopt as many protocols as possible for the proposed architecture. We do not directly compare the selected protocols in order to show which is good or bad from the architectural perspective. Instead of such comparisons, we will show that the selected protocols can be used for the Mobile IPTV system architecture such that the system performance satisfies the minimum IPTV QoS/QoE requirement defined by ITU-T [11].

Our contributions are mainly as follows:

- A System Architecture for Mobile IPTV Services.
- Seamless Vertical Handover Schemes between Heterogeneous Networks.
- Signaling extension to RTSP and SCTP for the proposed architecture based on Mobile IPTV service modeling.

#### 4.1. Mobile IPTV service formulation

Mobile IPTV architecture should allow service continuity over heterogeneous networks including wired and wireless. Therefore, handover (i.e., both vertical and or horizontal handover) occurs when a mobile device is moving around. In this section, we formulate the IPTV content forwarding in mobile networks as follows: Given a mobile network with Point of Attachments (PoAs), such as Access Points (APs) and Base Stations (BSes), our goal is to deliver IPTV contents packets reliably from the contents server to a destination mobile device with a minimum E2E delay.

First, we explain how to select an optimal target point for the IPTV content delivery from the content server to a

destination mobile device with the packet delay and movement delay distributions. The target point selection is performed on the basis of the delivery probability that the packet will arrive earlier than the destination device at the target point that is a rendezvous point of the packet and the destination device. This delivery probability can be computed with the packets delivery delay distribution and the destination devices travel delay distribution as follows:

Let  $PoA$  be the set of PoAs along the trajectory of the user's mobile device. Let  $i$  be the target point candidate as PoA for IPTV service where  $i \in PoA$ . Let  $\alpha$  be the user defined delivery probability. Let  $S_i$  be the IPTV service delay from the contents server to the target point  $i$ . Let  $U_i$  be the user's travel delay from its current position to target point  $i$ . Let  $f(S_i)$  be the probability density function (PDF) of the IPTV service delay  $S_i$ . Let  $g(U_i)$  be the PDF of the user's travel delay  $U_i$ . It is assumed that the user's trajectory is available by either the user's historical mobility trace data or an explicit navigation path through a smartphone navigator App (e.g., Waze [12] and Navfree [13]) with the user's destination and navigation path by the user's input. In this section, the delay distributions of  $f(S_i)$  and  $g(U_i)$  will be modeled as Gamma distributions with the means and standard deviations of the IPTV service delay and the user's walking speed [14].

For example, in Fig. 2,  $S_{10}$  is the expected service delay that the IPTV contents will be delivered from the contents server to the target point  $n_{10}$  and  $U_{10}$  is the expected user travel delay from the mobile device's current position to the target point  $n_{10}$ . Thus, we can compute the delivery probability as  $P[S_i \leq U_i]$ ; note that  $P[S_i \leq U_i]$  will be formally defined in (4). With this delivery probability, we can perform the target point selection to choose an optimal PoA among PoAs along the moving path of the mobile device as a temporary packet buffer node to keep packets for the mobile device.

Given the user-required delivery probability threshold  $\alpha$  (e.g., 0.99), we select a target point intersection  $i$  with the minimum user travel delay as an optimal target point such that  $P[S_i \leq U_i] \geq \alpha$ . We formulate the selection of a target point selection  $i^*$  as follows:

$$i^* \leftarrow \min_{i \in PoA} E[U_i] \quad \text{subject to } P[S_i \leq U_i] \geq \alpha. \quad (2)$$

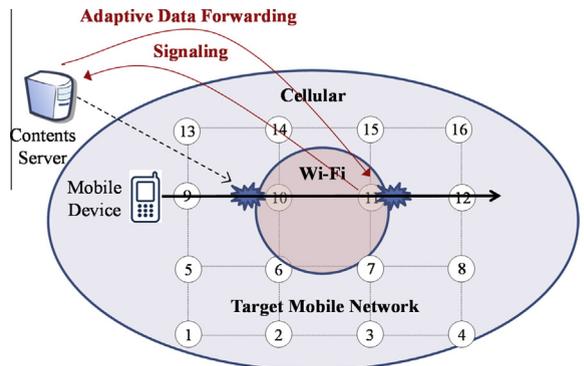


Fig. 2. IPTV packet forwarding in mobile network.

Actually, the minimum user travel delay determines the destination user's packet reception delay. More formally, we can select an optimal target point with a minimum delivery delay while satisfying the user defined delivery probability  $\alpha$  as follows: In (2), the delivery probability  $P[S_i \leq U_i]$  is the probability that the IPTV packet will arrive earlier at target point  $i$  than the destination user [15]. Fig. 3 shows the distribution of IPTV service delay  $S$  and the distribution of user travel delay  $U$ . We model the distributions of IPTV service delay and user travel delay as the Gamma distributions [14] such that

$$U \sim \Gamma(\kappa, \theta), \quad (3)$$

where the average walking speed of user is 1.7 m/s per PoA-to-PoA (e.g., PoA denoted as 10 and PoA denoted as 11 in Fig. 2) with  $E[U] = 10.5$  s and  $STD[U] = 2.1$  s. In our service model, we do only consider the walking speed of user with a mobile device for the architecture evaluation and do not consider other moving situations, such as vehicular networks since the major goal of this paper is to prove the idea of the new architecture rather than to perform mobility comparisons.

Our delay models are not restricted to the Gamma distributions and can accommodate any empirical distributions. That is, if more accurate distributions are available, our model can use them for the computation of the delivery probability, which will be described in (4). Given that the service delay distribution and the user travel delay distribution are independent of each other, the delivery probability  $P[S_i \leq U_i]$  is computed as follows:

$$P[S_i \leq U_i] = \int_0^{TTL} \int_0^u f(s)g(u)dsdu, \quad (4)$$

where  $f(s)$  is the probability density function (PDF) of the IPTV service delay  $s$  and  $g(u)$  is the truncated PDF of the user travel delay  $u$  with the integration upper bound, and TTL that is the Time-To-Live as service datas lifetime. Note that the delivery probability is computed considering the packets lifetime TTL; that is, since the packet is discarded after TTL, the probability portion is zero after TTL. Clearly, the optimal target point selection depends on the service delay model  $S$  and the user delay model  $U$ . The PDF and CDF of user travel delay are depicted in Fig. 4.

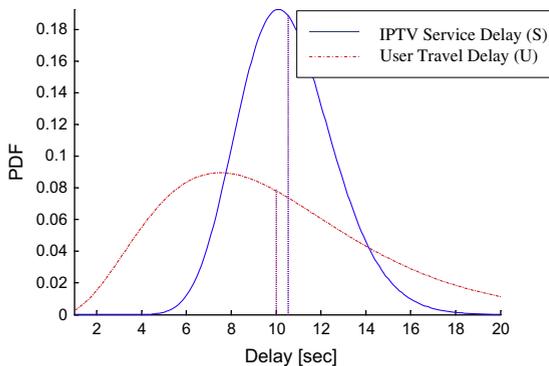


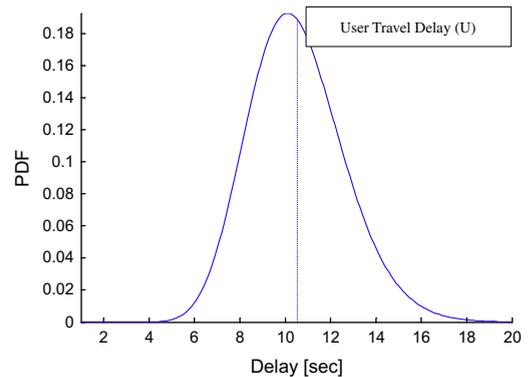
Fig. 3. The distributions of IPTV service delay and user travel delay.

For the handover signaling, we can use two approaches such as reactive signaling and proactive signaling. The reactive signaling in Fig. 2 does not cover the mobile user's trajectory prediction. Once handover occurs in heterogeneous access networks, a signaling is triggered and then sent to the contents server from the mobile user's device. However, the mobility of the user's device is not considered for signaling.

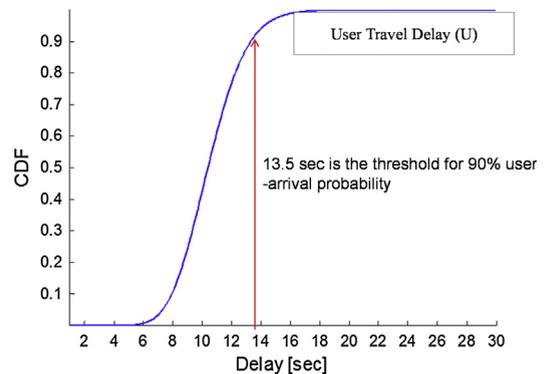
Meanwhile, proactive signaling predicts the mobility of the user's device (so called trajectory prediction for user) and forward packets to the target points selected by (2)–(4) before user reaches the destination. For the illustration, our proactive signaling is conceptually depicted in Fig. 5. Although the proactive signaling is highly appropriated for the user mobility, it requires various delay models and mobility studies deeply. This proactive signaling is not a major scope of this paper, but we illustrate it as an enhanced signaling for our Mobile IPTV architecture. In this paper, for the walking users with mobile devices, we currently use the reactive signaling for the proposed system architecture. We will implement the proactive signaling for vehicles moving fast in the road network as future work.

#### 4.2. Handover decision for heterogeneous access networks

Vertical handover decision algorithms are essential components of the architecture of the heterogeneous



(a) The PDF of user travel delay



(b) The CDF of user travel delay

Fig. 4. The PDF and CDF of user travel delay.

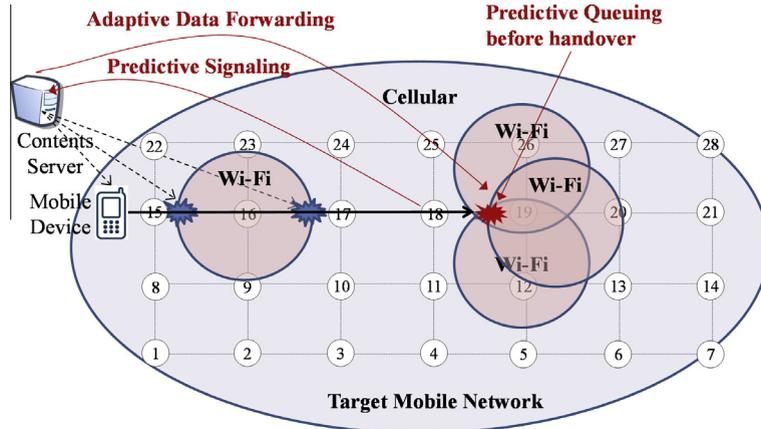


Fig. 5. Proactive signaling model for Mobile IPTV.

access networks. These algorithms need to be designed in order to provide the required QoS and QoE to the seamless IPTV service while using the different network interfaces. A number of previous studies have surveyed vertical handover decision algorithms such as received signaling strength (RSS) based algorithm, bandwidth based algorithm, cost function based algorithm, and various combination algorithm [16]. Practically, RSS based algorithm is quite popular because of the simplicity of the hardware required for RSS measurements. It compares the RSS of the current PoA with the others to make handover decisions. In the case where the new RSS is better than the current RSS, then the handover decision is made by the mobile device. Otherwise, the mobile device is maintaining its connection to the current PoA and does not make the handover decision. This algorithm does not fit well for the proposed architecture since the handover decision must be triggered whenever the new network interface comes up on the mobile device. This is because in our architecture, the come-up of the new network interface happens only when the new network interface can provide the mobile device with more reliable network connectivity for the Mobile IPTV service.

Meanwhile, there exists a bandwidth-based vertical handover decision algorithm, considering available bandwidth for the mobile device or traffic demand. Practically, this algorithm compares both the current network available bandwidth and the new attached network bandwidth. For delay-sensitive applications, a handover occurs only if the current serving network is not able to provide enough bandwidth for the application while the new attached network is able to provide the required bandwidth. In [17], the approximate value of the residual bandwidth is evaluated by (5):

$$B_r = T \times (1 - \alpha \times U_c) \times (1 - \gamma), \quad (5)$$

where  $B_r$  is residual bandwidth,  $T$  is throughput,  $U_c$  is channel utilization,  $\gamma$  is packet loss rate, and  $\alpha$  is a factor reflecting the network link characteristics (e.g., IEEE 802.11 MAC overhead as set to 1.25).

Although this algorithm is closely related to the targeted IPTV application in the time-sensitivity aspect, it may

does not work well in the proposed architecture because the handover decision happens regardless of the available bandwidth in the proposed architecture. Rather, the handover decision occurs whenever the new network is available even though this new network bandwidth is lower than the current network bandwidth. As in the case of RSS, in our architecture, the come-up of the new network interface occurs only when the new network interface can provide the mobile device with more reliable network connectivity for the Mobile IPTV service.

Therefore, we choose the cost function based vertical handover decision algorithm combining metrics in a cost function. We evaluate our proposed system architecture using bandwidth and stabilization time factors according to the IPTV QoE requirements. Thus, both bandwidth and stabilization time are used for the cost function factors when deciding the handover time.

In [18], vertical handover decision algorithm relies on a cost function which calculates the cost of possible target networks. The algorithm prioritizes all the active applications, and then the cost of each possible target network for the service with the highest priority. We adopt this algorithm with our own modification according to the requirements of our proposed system architecture. The per service cost  $C_{iptv}^n$  by network  $n$  for IPTV is calculated as follows:

$$C_{iptv}^n = \sum_{j=1}^m W_{iptv,j}^n Q_{iptv,j}^n, \quad (6)$$

where  $Q_{iptv,j}^n$  is the normalized QoS provided by parameter  $j = 1, \dots, m$  and  $W_{iptv,j}^n$  is the weight indicating the impact of the QoS parameter  $j$  on the user or the network. The total cost is the sum of the cost of each QoS parameter, including bandwidth and stabilization time. The service is provided by the network with the minimum cost. If the next network candidate cost is less than the current network cost, then the handover decision is made, otherwise the mobile device keeps working on the current network interface without any handover.

In our evaluation, we set the weight parameter  $W_{iptv,j}^n$  to 0 since it is considered whether to perform the handover

whenever the new network is attached regardless of bandwidth and link condition. Therefore, the total cost  $C_{iptv}^{new}$  for the new network is the minimum cost in all handover cases. In case of  $n \geq 2$  which implies that at least two kinds of candidate networks are overlapped as depicted in Fig. 5, the mobile device should consider the best candidate network for the handover decision, considering the handover cost that is the sum of the products for the weighted QoS parameters in (6). This research topic is included in our future works for further study.

As described in Section 4.1, we choose reactive signaling for the current architecture and leave proactive signaling for the future work. Note that in the case of proactive signaling model, the candidate networks are able to calculate the total cost instead of the mobile device for preparing for the vertical handover. For that purpose, we illustrate on the weighted function based heuristic for the vertical handover decision algorithm [19].

The quality  $Q_i$  of the next network candidate  $i$  is defined as a linear combination of weights as follows:

$$Q_i = W_B B_i + \frac{2W_{MP}}{MP_i} + \frac{W_{UD}}{UD_i} + \frac{W_{SD}}{SD_i} + \frac{W_C}{C_i}, \quad (7)$$

where  $Q_i$  represents the quality of network  $i$ ,  $B_i$ ,  $MP_i$ ,  $UD_i$ ,  $SD_i$ , and  $C_i$  are bandwidth, moving probability, user travel delay, IPTV service delay, and monetary cost of service, respectively, and  $W_B$ ,  $W_{MP}$ ,  $W_{UD}$ ,  $W_{SD}$ , and  $W_C$  are their respective weights. Thus, the bandwidth is linearly proportional to the quality of network. The wider bandwidth is better for the quality of network in (7). Otherwise, other cost parameters are reciprocally proportional to the quality of network in (7) with the following equality:

$$W_B + W_{MP} + W_{UD} + W_{SD} + W_C = 1. \quad (8)$$

As depicted in Fig. 5, the delivery probability  $P[S_i \leq U_i]$  in (4) is the probability that the IPTV packet will arrive earlier at target point  $i$  than the destination user. Therefore, we have:

$$W_{SD} \gg W_{UD}. \quad (9)$$

Regarding the monetary cost of service, user subscription cost should be considered for the QoS  $Q_i$ . In case of  $n = 3$ , where Wi-Fi, WiMAX, and 3G heterogeneous networks are deployed, the weighted cost is calculated by

$$W_{3G} > W_{WiMAX} > W_{Wi-Fi}. \quad (10)$$

But the order of the weighted costs can be changed according to the network configuration and conditions. The candidate network with the highest  $Q_i$  is selected as the handover target network. By letting the calculation done by the visited network, the resource of the mobile device can be saved so that the visited network system is able to achieve short handover decision-making delay, low handover blocking rate and high throughput. However, the method requires extra cooperation between the mobile device and the PoA of the visited network, which may cause additional latency and excessive load to the network when there are a large number of mobile devices. In next section, we will evaluate our proposed Mobile IPTV system architecture.

## 5. Evaluation criteria

For the validation of the proposed architecture, we evaluate it by network simulation. For this evaluation, we should propose an appropriate criteria and the relevant information is described in this section.

From the IPTV perspective, stabilization time is a very sensitive factor for the quality of user experience while watching IPTV [2]. This stabilization time can work as a major point of satisfaction. In particular, how quickly and correctly the subscribers can change channels is an important part of IPTV QoE, and this technical factor (a.k.a. channel zapping) is closely related to the stabilization time. In terms of user perception during the handover, several factors such as delay, jitter, and frame rate are significant. Particularly, these factors are closely related to stabilization time, and impact user's QoE. The results in [20] show that a significant reduction in frame rate does not reduce the user's understanding of the information. On the other hand, the delay and jitter make more impact on the user's QoE. The recommended QoE requirements are measured by data throughput and stabilization time in our evaluation. It is because these performance metrics mainly contribute to QoE of the media stream in IPTV as illustrated in [11].

Bit rate basically means the number of bits of data that can be transported over the networks. In the case of compressed video like IPTV contents there are two primary processing activities for which the term of throughput is often applied, such as encoding and decoding. Data throughput means how fast that encoding or decoding process can occur. Therefore, we choose data throughput as a performance metric instead of simple bit rate in our evaluation. Practically, the value of data throughput is greater than the real value of bit rate. The performance requirements for IPTV video service in ITU-T is described in Table 2. Actually, the minimum requirement of data throughput is around 7 Mbps for delivering HD video and the 2–3 Mbps for the SD video. In addition, acceptable channel zapping delay is generally considered to be around 1 s for the E2E path from the server to the client. The channel zapping time of 100–200 ms is considered instantaneous by viewers [21]. To ensure the transfer of information, both delay and jitter should be adapted to maintain the user's level of enjoyment. The evaluated delay in this paper is acceptable for IPTV user. Thus, our evaluation criteria are as follows:

- Data\_throughput  $\geq 2$ –3 Mbps for SD video delivery.
- Data\_throughput  $\geq 7$  Mbps for HD video delivery.
- Stabilization\_time  $< 200$  ms.

These criteria will be applied for the system evaluation in this section.

To validate our proposed architecture, we evaluate it with INET framework in OMNeT++ network simulator [22] since it supports discrete event simulation for IP-based networks. In addition, both SCTP and RTSP used for the evaluation in this paper are well defined in INET including the simulation model and parameters. INET

framework also supports external interfaces that allow hybrid scenarios where simulated nodes communicate with real external IP-based nodes.

The network topology for the evaluation consists of one contents server and three network interfaces and configuration parameters are described in Table 1 in detail. The wired links between contents server and each wireless network are configured to have a link capacity of 1 Gbps and the wireless channel between the Mobile IPTV client and Access Point (AP) is defined to a data rate of 100 Mbps. Because, data channel is shared by numerous Mobile IPTV clients in the real wireless network, we set up the wireless network consisting of an AP and the corresponding Mobile IPTV clients. 100 Mobile IPTV clients are deployed in each area of wireless network. Mobile IPTV client and contents server are connected to each other via an 802.11b AP.

When the simulation starts, firstly each Mobile IPTV client is randomly chosen to turn on its IPTV service and also an IP address on the Mobile IPTV client is dynamically configured by an external DHCP server. Then Mobile IPTV client requests an IPTV service to contents server. This IPTV service generates background traffic as constant bit rate traffic with almost realistic characteristics in the network environments.

Note that this paper focuses on the performance comparison between the existing solution and our proposed solutions using either RTSP or SCTP for the Mobile IPTV system architecture. It will be also good to investigate whether our proposed solutions outperform other related solutions, particularly HTTP-based solution [5] described in Section 3, and we will compare our proposed solutions with other related solutions as future work. In next subsections, we will evaluate our two approaches for Mobile IPTV service.

## 6. Application layer evaluation for Mobile IPTV

As depicted in Fig. 1, Mobile IPTV architecture is doable with various protocol layers. This paper uses both application and transport layer for the performance evaluations. In addition, this paper configures hybrid experimental environments of heterogeneous access networks such as Wi-Fi and WiMAX for RTSP and Wi-Fi and Ethernet for SCTP.

In this section, we describe the RTSP-based approach for Mobile IPTV as an application layer evaluation. Based on the selected requirements in Table 2, RTSP is used for the

signaling in this section as application layer architecture. RTSP is currently most popular for providing real-time multimedia streaming over the networks, and fits well for the architecture design according to the market requirements.

### 6.1. RTSP-based approach

To provide the function of adaptive sending control in Mobile IPTV, we choose RTSP with several advantages. As a complementary way, a quick-start TCP algorithm [23] is combined in order to provide faster sending rates than the slow-start algorithm. It is because the adaptive sending control feature is tightly coupled to the transmission layer either UDP or TCP. Despite RTSP over UDP, we tried to optimize RTSP operation over TCP since TCP is used as a transport protocol for the implementation of stable IPTV service. In fact, TCP can prevent packet loss at application layer due to its connection-oriented feature, but also cause long delay under such loss events and a significant impact on QoE [11]. For the validation of the proposed mechanism in TCP in terms of time delay, we evaluated stabilization time in conjunction with data throughput. During handover, data communications between the mobile device and the network are unstable and shaky for a while and going to be settled with a stable status after few seconds. Then, bit rate is going to be increased continuously. The gap between the handover initiation and the starting point of stable bit rate is defined here as stabilization time.

TCPs slow-start algorithm controls the sending rate using a congestion window (denoted as  $cwnd$ ), which limits the amount of data that can be transmitted to the network before receiving an acknowledgement. This algorithm requires a significant number of round-trip times (RTT) and large amount of data to open  $cwnd$  and efficiently use available bandwidth. The quick-start design alleviates the slow-start delay for connections in under-utilized environments like heterogeneous networks. The determination of an appropriate sending rate is fundamental to the function of IP networks including multiple IPTV networks. The TCP sender computes its quick start congestion window by

$$cwnd = \frac{Rate \times RTT}{MSS + H}, \quad (11)$$

where  $Rate$  is the approved rate request in bps,  $RTT$  is the recently measured round-trip time in seconds,  $MSS$  is the maximum segment size for the TCP connection in bytes, and  $H$  is the estimated connection header overhead in bytes. However, the quick-start algorithm requires support from all of the routers along a path. Additionally, the most appropriate method for determining the data rate (i.e.,  $Rate$  in (11)) according to network characteristics remains an unresolved question. Nevertheless, the quick-start TCP is a good algorithm when link optimization is low and network changes happen frequently over heterogeneous networks, and. Thus, this quick-start TCP well fits well for Mobile IPTV.

### 6.2. Detailed operation

The key points in the proposed mechanism are to deal with the two important parameters as Speed header in

**Table 1**  
Configuration parameters in performance evaluation.

Network parameters			
Network type	Wi-Fi	WiMAX	Ethernet
Frequency	2.4 GHz	2.4 GHz	–
Data rate	4.5–11 Mbps	19–54 Mbps	100 Mbps
Range	115 feet	124 feet	–
IPTV Contents Server Parameters			
Service	Bandwidth	Period	Payload
HDTV	20 Mbps	1 ms	2500 Bytes
IPTV Mobile Device Parameters			
Network interface	Ethernet, Wi-Fi (802.11b), WiMAX		
Mobility type	Linear mobility		
Mobility speed	3–6 km/h (walking speed)		

**Table 2**  
Performance requirements for IPTV video service in ITU-T standard.

Source type	Video codec standard (Non-inclusive list)	Minimum bit rate (Video elementary stream level)
SD broadcast program sources	H.262-Main profile at main level (MP@ML) H.264 (Main profile at level 3.0), SMPTE 421, AVS	2.5 Mbit/s CBR 1.75 Mbit/s CBR
H.262 SD, VoD and premium program sources	H.262-Main profile at main level (MP@ML) H.264 (Main profile at level 3.0), SMPTE 421, AVS	3.18 Mbit/s CBR 2.1 Mbit/s CBR
HD broadcast program sources	H.262-Main profile at main level (MP@ML) H.264 (Main profile at level 3.0), SMPTE 421, AVS	15 Mbit/s CBR 10 Mbit/s CBR

RTSP and Rate in TCP according to the changeable network conditions, where the type of network connection is used for detecting an appropriate sending rate in conjunction with two parameters. It is because RTSP is typically running on TCP connection and tightly integrated each other in the real implementation. It can be replaced with appropriate ones that confirm to the requirements required by a specific IPTV service provider.

Fig. 6 shows the operation steps between the Mobile IPTV client and the IPTV contents server. Once network properties are obtained via the Network Driver System (Steps 1 and 2), Speed value in RTSP stack and Rate parameter in TCP stack are set according to network type (Steps 3 and 4). When the network type is identified as the IPTV client, Speed request-header and TCPs quick-start request are triggered in order to immediately inform the server of the client condition changes (Steps 5 through 8). The transmission rate is then adjusted accordingly (Step 9). The detailed work flow the mechanism is as follows:

The first work is to control Speed header in RTSP. Speed request-header field instructs the server to deliver specific amounts of nominal media time per unit of delivery time, contingent upon the servers ability and desire to serve the multimedia stream at the given speed. Speed header determines the bandwidth used for data delivery and is meant for use in specific circumstances, such as when higher or lower rate delivery of a presentation is desired. The server is able to indicate its level of support through a feature-tag (play.speed) and Speed parameter values are expressed as positive decimal values. The Speed value

of zero is invalid, and the range is specified in the form of (lower\_bound,upper\_bound).

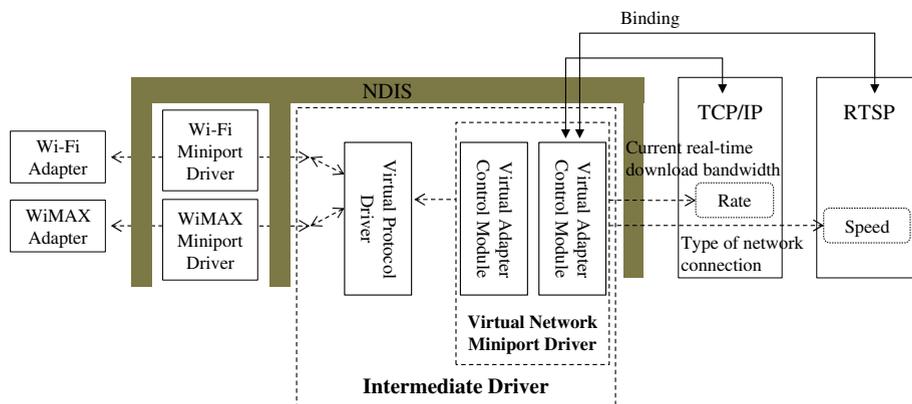
We conducted a functional experiment to test the efficacy of our mechanism. For this implementation, Speed value range is specified as 0.5–2 where 1 is normal speed, 0.5 is low speed, and 2 is high speed. Speed value is a decimal number and mapped to a specific parameter in Mbps such as speed\_value (2) = 15 Mbps, which means that speed value 2 is mapped to 15 Mbps. In order to determine Speed value, the type of network connection obtained by the new Network Driver System as depicted in Fig. 6 is used within the Mobile IPTV client whenever the new network interface is detected dynamically as follows:

- network\_type:ethernet = speed\_value (2),
- network\_type:wifi = speed\_value (1), and
- network\_type:wimax = speed\_value (0.5).

These network properties are extracted by the new Network Driver System to trigger quick-start request and identify RTSP speed-request header with an appropriate sending rate when multiple network interfaces are equipped within the IPTV client.

The second work is to control Rate parameter in TCP. In order to quickly adjust the data rate within TCP, Rate parameter is dynamically mapped to Speed parameter as follows:

- speed\_value (2) = 15 Mbps,
- speed\_value (1) = 7.5 Mbps, and
- speed\_value (0.5) = 2.5 Mbps.



**Fig. 6.** Network driver system for RTSP-based mechanism.

The defined value of the Rate parameter is based on the ITU-T definition [11]. These requirements are for user acceptability of the IPTV services from QoE perspective. The recommended minimum application layer performance can be measured by data throughput and stabilization time used to evaluate the experimental results.

The third work is to calculate the appropriate sending rate based on the obtained Speed and Rate parameters in the client side. In TCP, the congestion window is calculated by (11), and initiates the transport session. In RTSP, the obtained Speed parameter is sent to the server using Speed request-header to request a particular sending rate. The implementation of sending rate changes depends on the server, and we implement the Speed parameter features in the IPTV contents server according to the ITU-T definition as follows:

- speed\_value (2) = 15 Mbps,
- speed\_value (1) = 7.5 Mbps, and
- speed\_value (0.5) = 2.5 Mbps.

The last work is to adjust the sending rate in the server side according to the RTSP and TCP connections initiated by the client for the new IPTV session. The practical sending rate in the server is subject to change based on the TCP congestion window since the calculated window size can be less than the initiated sending rate from RTSP. In order to trigger Speed parameter in RTSP and Rate parameter in TCP quick-start, the information about network type must be sensed immediately when the network type is changed. For the Network Driver System, we adapted the Windows operating system NDIS [24] and the W3C Device Application Programming Interface (API) specification [25] as an intermediate driver to support the network switch between Wi-Fi and WiMAX as heterogeneous network. The NDIS is used to communicate with network card drivers and to support basic services such as allowing a protocol module to send raw packets over a network device and to be notified of incoming packets received by a network device. As depicted in Fig. 6, the Network Driver System lies between the legacy protocol driver and the lower adapter driver. It behaves like a lower adapter driver from the perspective of the upper level transport driver, and like a protocol driver from the perspective of the lower adapter driver.

As soon as the new interface is switched by the intermediate driver, the type of network connection for Speed parameter in RTSP and Rate parameter in TCP are exposed and delivered to the higher protocol as TCP and RTSP immediately.

### 6.3. Performance evaluation

Our experimental implementation, the mobile device is used for manipulating Speed parameter in RTSP and Rate in TCP quick-start via multiple wireless interfaces, such as Wi-Fi and WiMAX, within the Mobile IPTV client. The initial transmission rate of the IPTV contents server remains unchanged until the Speed and Rate parameters are obtained from the mobile device. In addition, Table 2 shows stabilization time from the case in which the mobile device moves from a high-bandwidth network to a low-bandwidth network and vice versa. Bit rate equals the amount of data received for stabilization time where the average walking speed of user is 1.7 m/s based on the service modeling in Section 4.1.

We assessed data throughput and stabilization when the mobile device moved from Wi-Fi to WiMAX and from WiMAX to Wi-Fi and measured a total of ten times for two different cases as follows:

- A manipulating network condition without Speed and Rate parameter notification.
- A manipulating network condition after transmitting Speed and Rate parameter values.

In particular, data throughput increases to 7.64 Mbit/s, and 2.32 Mbit/s, as shown in Table 3 and Fig. 7. Both resulting data throughput approximately satisfy the minimum requirements of the IPTV services in heterogeneous networks from bit rate perspective since these values include encoding/decoding process. Therefore, the actual bit rate without encoding/decoding is not less than the minimum requirements of ITU-T.

Also, acceptable channel zapping delay is generally considered to be total around 1 s in the E2E path. A channel zapping time of 100–200 ms is considered instantaneous by viewers [21]. Consequently, all stabilization time, 233.24 ms and 9.21 ms, measured by the proposed mechanism is in scope of acceptable delay when network interface is changed. The results are shown in Fig. 8.

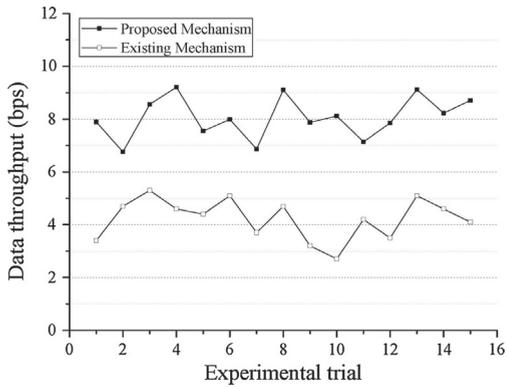
We used Vovida SIP-1.5.0, which is comprised of SIP, RTP, RTCP and RTSP within the Linux kernel, version 2.6.20.11. In [26], the Linux kernel has been extended in order to support TCPs quick-start extension. Our experimental implementation, within a real TCP/IP stack, allows us to test quick-start in combination with real applications within a real network.

### 6.4. Summary

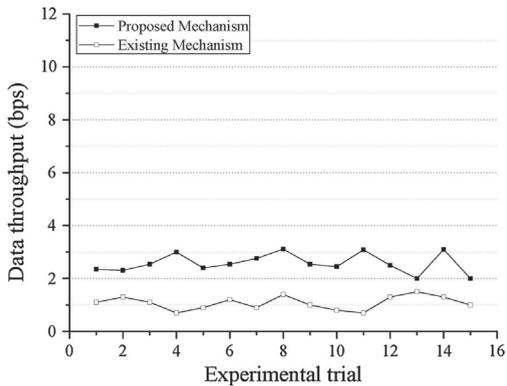
We proposed the RTSP-based adaptive sending control according to the network conditions, and implemented it

**Table 3**  
Evaluation results with RTSP in the proposed architecture.

Measurement	WiMAX to Wi-Fi		Wi-Fi to WiMAX	
	Existing scheme	Proposed scheme	Existing scheme	Proposed scheme
Data throughput (bps)	4.39 M	7.64 M	1.05 M	2.32 M
Stabilization time (ms)	636.34	233.24	78.78	9.21



(a) Data throughput by switching from WiMAX to Wi-Fi



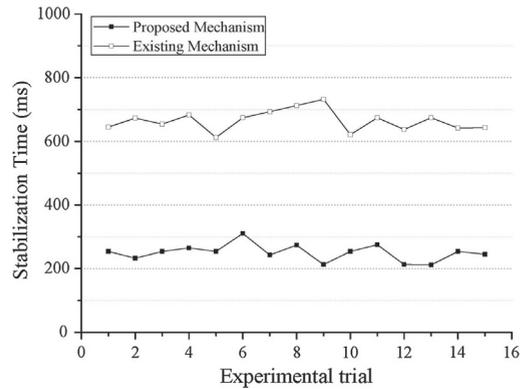
(b) Data throughput by switching from Wi-Fi to WiMAX

**Fig. 7.** Evaluation result of data throughput by switching between WiMAX and Wi-Fi.

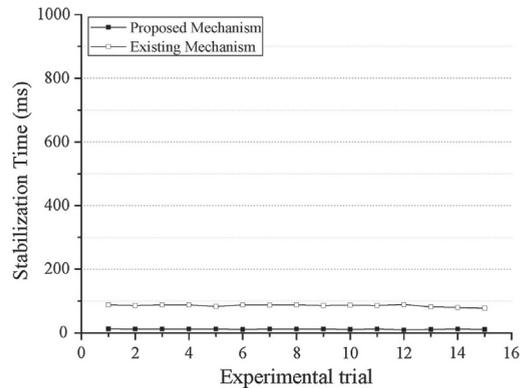
in Mobile IPTV system. The key points in the proposed mechanism have been to deal with the two important parameters as Speed header in RTSP and Rate in TCP according to the changeable network conditions, where the type of network connection was used for detecting an appropriate sending rate in conjunction with two parameters. It was because RTSP was typically running on TCP connection and tightly integrated each other in the real implementation. It can be replaced with appropriate ones that confirm to the requirements required by a specific IPTV service provider. The result has shown that the proposed mechanism met the requirements of the minimum application layer performance recommended by ITU-T for the quality of experience over heterogeneous networks.

## 7. Transport layer evaluation for Mobile IPTV

Currently, many proposals have been made for performing handover while roaming across heterogeneous networks. These approaches operate at different layers of the network protocol stack. When designing a new architecture for implementing vertical handover, it is important to limit the modifications required to existing systems, and to minimize the amount of network traffic needed. In [27], several issues of which layer in the Internet protocol stack mobility belongs to are well addressed. They address the



(a) Stabilization time by switching from WiMAX to Wi-Fi



(b) Stabilization time by switching from Wi-Fi to WiMAX

**Fig. 8.** Evaluation result of stabilization by switching between WiMAX and Wi-Fi.

various strengths and weaknesses of implementing mobility at three different layers of the protocol stack, and conclude that the transport layer is the most likely place for a mobility protocol, but the best approach may be a cross-layer approach where inter-layer communications is used.

Typically, TCP and UDP are widely used for the transport layer in the Internet. To support more functionality on the transport layer against both TCP and UDP, a couple of transport protocols as SCTP and DCCP are proposed and standardized by IETF recently. Based on these protocols, we try to study on them and extended them for signaling the proposed Mobile IPTV architecture. For supporting the seamless IPTV service in the proposed architecture, mobility across different heterogeneous access networks are absolutely required. To implement mobility at the transport layer, there must first be means for a host to detect new networks that it moves to, and obtain new IP addresses in them. After new network information is obtained, the transport layer bindings at remote hosts must be updated for existing connections. In this section, SCTP has its own mobility features for the proposed architecture.

### 7.1. SCTP-based approach

For real implementations with heterogeneous access networks, we chose three access network interfaces:

Ethernet high bandwidth connection, Wi-Fi middle bandwidth connection, and WiMAX low bandwidth connection. SCTP was used at the transport layer with new options which carry INC to the IPTV contents server from the mobile device in the proposed system architecture. SCTP is a new reliable transport layer protocol. Compared to TCP, SCTP provides the distinctive features such as multi-streaming and multi-homing.

Especially, SCTP multi-streaming feature enables an upper layer application to separate the data streams logically by using the Sockets API [28] with different stream identifiers (SIDs) for each of the user data streams. In addition, SCTP can be used to reduce the well-known Head-of-Line blocking effect that occurs in the TCP connection. In Fig. 9, an SCTP session is established between the IPTV mobile device and the contents server over the legacy network. SCTP lies between the network layer and the application layer and all application data is passed through the APIs.

In order to recognize what network is currently active in application layer, NDIS [24] is also used likewise in RTPS evaluation. It behaves like a lower adapter driver from the perspective of the upper level transport driver, and like a protocol driver from the perspective of the lower adapter driver. The intermediate driver supports 1 (virtual network miniport): 3 (LAN, WLAN and WiMAX) multiplexing. In the IP implementations, the outgoing interface of a multiple interface host is often determined by the destination IP address. The mapping of outgoing source IP address and destination address is done by a lookup in the host routing table maintained by the operating system and experimentally bound to the virtual network miniport in the

Windows operation system. This configuration arises because of the multi-homing feature of SCTP in the IPTV mobile device.

### 7.2. Detailed operation

To piggyback INC to the server, ASCONF [29] is applied for the proposed IPTV system. When a local SCTP client has multiple points of attachment to the Internet, ASCONF allows an SCTP stack to dynamically add an IP Addresses to an SCTP association, dynamically delete an IP addresses from an SCTP association, and to request to set the primary address the peer will use when sending to an endpoint.

In performing ASCONF procedure, INC is delivered to the IPTV contents server by using a new INC option for piggybacking the network interface changes as well as network characteristics information. Using the multi-homing feature of SCTP, the Mobile IPTV client can have multiple IP addresses. Similarly, the IPTV contents server can also be configured for either single-homing (the server only one IP address to support handover) or multi-homing (the server allows more than one, usually two, IP addresses to support handover). In this Chapter, the single-homing configuration for the IPTV contents server and the two IP addresses configuration for the Mobile IPTV client are used for the experimental evaluation.

Fig. 9 shows the ASCONF procedure with the new INC option when the Mobile IPTV client moves to another network interface. To carry out INC, we defined the new INC option as depicted in Fig. 9 (chunk type (0xC1) is ASCONF). The information carried in the ASCONF chunk uses the form of a Type-Length-Value, and a new type as 0xC010

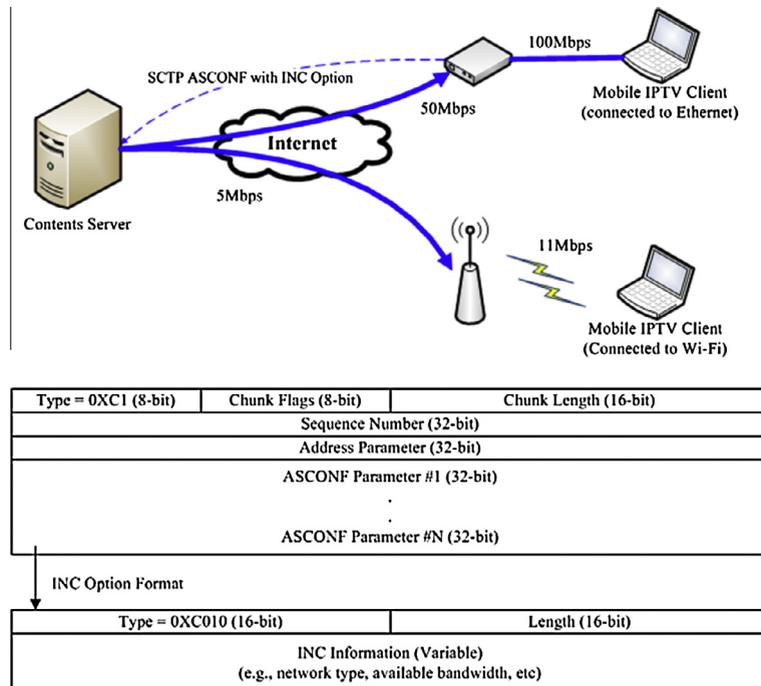


Fig. 9. System architecture and a new INC option in SCTP.

is defined for the INC option. Further field details can be found in [6]. For the IP address configuration while changing the network interface on the Mobile IPTV client, the handover procedure has the following three steps:

- **Step 1:** Add an IP address to an SCTP association.
- **Step 2:** Switch a network interface (during the handover).
- **Step 3:** Delete an IP address from an SCTP association.

To support the proposed architecture, two new steps are added in the form of INC option delivery in ASCONF message exchange, and adaptive streaming according to INC on the server side. Fig. 10 shows the handover procedure according to the extended five steps accordingly. Given the multi-homing feature of SCTP, the Mobile IPTV client with multiple interfaces can have easy and efficient handover solution for mobility support over IP networks since the multi-homing mechanism is originally intended to be used for fault-resilient communications between two SCTP endpoints.

To support IP mobility using SCTP, mobile SCTP was developed without any support of network routers. It provides a new mechanism for dealing with IP add and change during the handover. It is, however not used for the proposed architecture in this Chapter. Instead, as described above, ASCONF in SCTP Dynamic Address Reconfiguration mechanism is chose for the proposed system architecture since it is an IETF standard solution.

In Fig. 10, the IPTV contents server is a single-homing meaning only one IP address configuration on the server and the Mobile IPTV client has two IP addresses for Wi-Fi and Ethernet interfaces. Whenever a new interface is enabled, INC is extracted by the Mobile IPTV client, and embedded into the new INC option within ASCONF message format to be sent to the IPTV contents server. Then, the new IP address of the enabled interface is informed to the server by using Add IP Address parameter. After the Mobile IPTV client receives an ASCONF\_ACK message

meaning the new IP address sent by the client is available, the IPTV mobile device sends an ASCONF message with Set Primary Address parameter to the server, and the data between the Mobile IPTV client and the server is routed through the primary IP address.

After switching the destination IP address of the Mobile IPTV client, the server is able to adjust its streaming volume according to INC sent by the mobile device. When the Mobile IPTV client comes back to its original interface, the IP address of the previous interface is released with Delete IP Address parameter within ASCONF message and disconnected accordingly. Fig. 10 shows the case of the handover from Ethernet to Wi-Fi, and the reverse case is exactly same procedure.

### 7.3. Performance evaluation

As shown in Fig. 10, the Mobile IPTV client has heterogeneous access network interfaces and at least one interface should be switched on. For the evaluation, we turn on Ethernet interface and send its INC to the IPTV contents server, then MPEG-2 streaming comes accordingly. After some time, we turn on another interface (either Wi-Fi or WiMAX), and switch the enabled interface, and send its INC by the new INC option in ASCONF message to the IPTV contents server. Then, the ongoing MPEG-2 streaming is adjusted according the selected interface. We presume handover occurs between different networks when in motion. For the evaluation, both Ethernet and Wi-Fi interfaces are enabled on the Mobile IPTV client.

In our implementation, the Mobile IPTV client and the IPTV contents server establish a session using the SCTP. The IPTV contents servers initial transmission rate is unchanged until the Mobile IPTV client notifies the server of its changed link status. To verify INC delivery's usefulness, we measured the data throughput  $T$  [bps] as follows:

$$T = N_{receive} \times \frac{S_{packet}}{D_{measure}}, \tag{12}$$

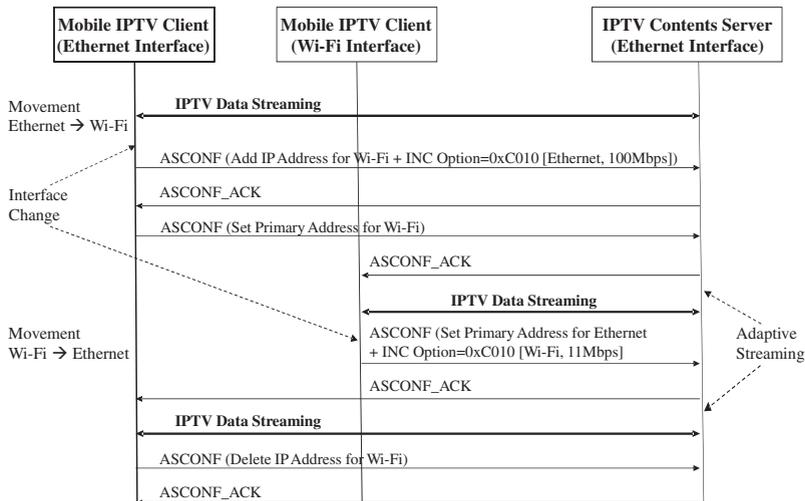
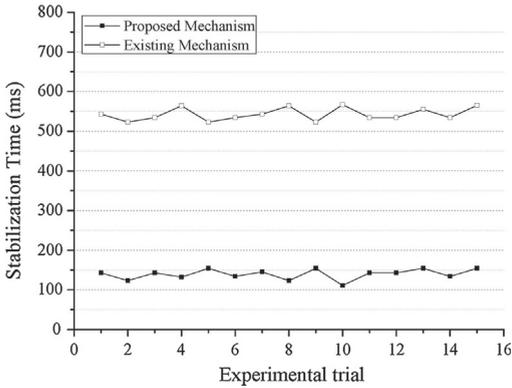


Fig. 10. Sctp association procedure.

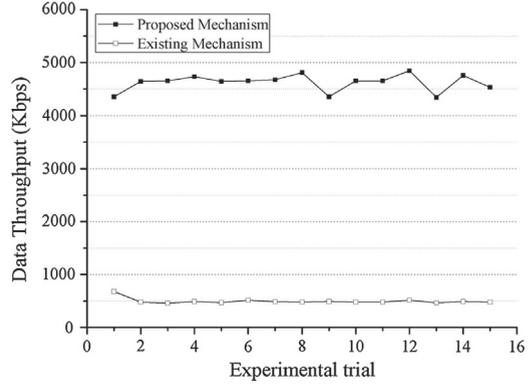
**Table 4**

Evaluation results with SCTP in the proposed architecture.

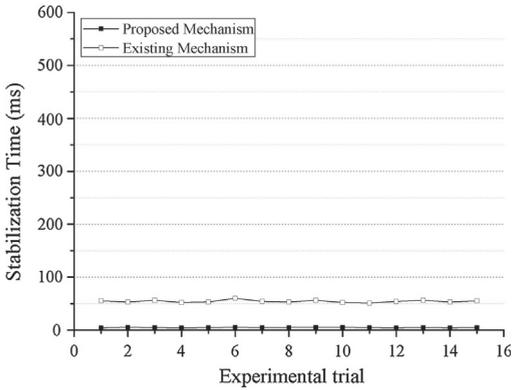
Measurement	Wi-Fi to Ethernet		Ethernet to Wi-Fi	
	Existing scheme	Proposed scheme	Existing scheme	Proposed scheme
Data throughput (bps)	324.32 K	49.67 M	923.43 K	4.56 M
Stabilization time (ms)	543.1	143.2	55.5	4.3



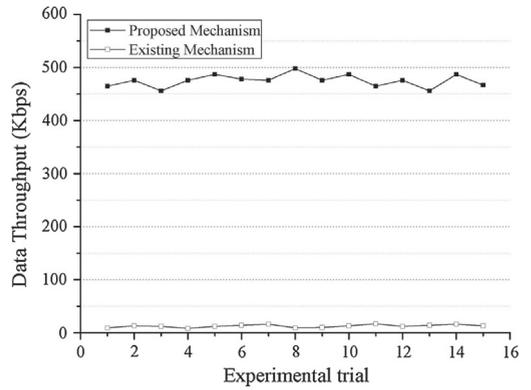
(a) Stabilization time by switching from Wi-Fi to Ethernet



(a) Data throughput by switching from WiMAX to Wi-Fi



(b) Stabilization time by switching from Ethernet to Wi-Fi



(b) Data throughput by switching from Wi-Fi to WiMAX

**Fig. 11.** Evaluation result of stabilization by switching between Wi-Fi and Ethernet.

**Fig. 12.** Evaluation result of data throughput by switching between Wi-Fi and Ethernet.

where  $N_{receive}$  is the number of packets received by the receiver,  $S_{packet}$  is the number of bits per packet [bits], and  $D_{measure}$  is measurement duration [sec]. We measured the data throughput parameters 30 times in two cases: handover without notification of INC and handover after transmitting INC.

We designed our IPTV service model, based on the currently most common commercial model with the current network configuration and the proposed system configuration for IPTV. Note that the specific network parameters are fully described in Table 1. Table 4 summarizes stabilization time and data throughput from a case in which the mobile device moves from a high-bandwidth network to a low-bandwidth network and vice versa.

Data throughput equals the amount of data received for stabilization time. During handover period, data

communications between the mobile device and the network is unstable and shaky for a while and going to be settled with a stable status after a few seconds. After this, the data throughput is going to increase continuously. The gap between the handover initiation and the starting point of stable data throughput is defined here as stabilization time. The data throughput measured using the proposed system architecture is efficient to resolve the problems of high packet loss and low data throughput in heterogeneous networks when the Mobile IPTV client is moving around. All results are depicted in Figs. 11 and 12.

In this evaluation comparison, the existing system in Table 4 means the current IPTV service working over heterogeneous wireless access networks using SCTP without the additional components. It is a general SCTP handover scenario between the different networks interfaces [30].

As described in Section 5, acceptable channel zapping delay is generally considered to be total around one second in the E2E path. A channel zapping time of 100–200 ms is considered instantaneous by viewers [11]. Consequently, all stabilization time, 143.2 ms and 4.3 ms, measured by the proposed system is within the scope of acceptable delay when the network interface is changed.

#### 7.4. Summary

We presented an SCTP-based system architecture that utilizes seamless IPTV services in heterogeneous access networks, signaling to dynamically adjust IPTV streaming sending rate in the scenarios where a user has the Mobile IPTV client and moves around.

## 8. Conclusion

Internet Protocol Television is defined as a multimedia service delivered over IP-based networks managed to support quality of service, quality of experience, security, interactivity, and reliability. This service is rapidly expanding to both wireless and mobile networks, and therefore enabling handover between heterogeneous access networks including homogeneous and heterogeneous networks is highly desirable to overcome both service-coverage limitations and eliminate dead spots.

This paper widely researches on the architectural aspect of Mobile IPTV to support QoS/QoE for seamless IPTV service in heterogeneous networks. Also, this paper proposes a new system architecture taking the selected implications into account. The results of all evaluations have shown that the proposed architecture met the requirements of the minimum IPTV service performance recommended by ITU-T international standard for the quality of experience over heterogeneous networks. In this paper, our Mobile IPTV system architecture has been evaluated on the reactive signaling. As future work, we will continuously research on the proactive signaling for our Mobile IPTV system architecture and also evaluate its performance along with enhanced vertical handover decision algorithms. We will also deeply research on the HTTP-based adaptive streaming and compare it with our proposed architecture, including the technical analysis of architectural advantages and disadvantages and also its service impact.

## Acknowledgment

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