

Seamless Streaming Media for Heterogeneous Mobile Networks

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Abstract As the mobile networking technologies evolve, people are able to access the Internet through heterogeneous wireless access networks, such as WLAN, GPRS, 3G and Beyond 3G networks. For the coverage, bandwidth and cost of these heterogeneous mobile access networks are quite different, a mobile host may hand over among them, and this is called vertical handoffs. One of the most important issues for heterogeneous mobile networks is that vertical handoffs may degrade the quality of the time-sensitive streaming media services, even interrupt them. To overcome the problem, in the paper a multicast-based redundant streaming architecture is proposed. The proposed architecture is implemented in the all-IPv6 heterogeneous mobile networks. Five experiments are performed to evaluate the performance of the proposed architecture. The experimental results and the analysis show that the proposed architecture is capable of providing seamless streaming services even if the vertical handoffs or the traffic congestion occurs. Moreover, it is found that the traffic overhead is only 1.0368% per vertical handoff for each mobile access network, and thus the feasibility of the proposed architecture is demonstrated.

This research was supported in part by National Science Council of the Republic of China under grant NSC 94-2213-E-008-018 and NSC 94-2219-E-260-006, and by Ministry of Economic Affairs under grant 95-EC-17-A-02-S1-029.

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Keywords multicast · seamless streaming ·
vertical handoff

1 Introduction

For mobile users want to be able to access the Internet ubiquitously, more and more mobile devices are equipped with multiple network interfaces connecting one or more access networks, so-called multihoming [1, 2]. There are many kinds of mobile access networks such as General Packet Radio Service (GPRS), 3G communications system, Beyond 3G communications systems, wireless LAN (WLAN), and Bluetooth. The bandwidth, coverage area and cost for the mobile access networks are quite different. The mobile access networks connected by a multihoming host play the role of backup for each other. Whenever a multihoming mobile host leaves the coverage of a mobile access network, the host will be handed over to another mobile access network, called vertical handoffs, so as to provide ubiquitous services [3].

In the heterogeneous mobile access network environment, how to decide the time for vertical handoffs and how to select an appropriate mobile network will affect deeply the quality of the provided services [4, 5]. Besides, vertical handoffs require additional time to reconnect to the new mobile access network, and the handoff time is larger than horizontal handoffs. The additional time for vertical handoffs always causes packet delay and losses, and thus degrades the quality of services (QoS) for streaming media. Therefore, how to provide smooth and seamless streaming media service becomes an important study issue. Solutions to multihoming can be classified into router-based and host-based approaches. Because of

the scalability problem [6], route-based approaches will not be considered in the paper. Host-based approaches can be implemented by host-centric approach [7], transport layer approach [8] and mobile approach [9]. However, the above solutions either introduce new routing protocols or modify existing routing protocols, and thus are hard to be deployed popularly.

In the paper, the multicast approach is adopted to realize seamless multihoming streaming media services, and any existing multicast routing protocols without any modification can be adopted, such as Multicast extensions to OSPF (MOSPF) [10], Protocol Independent Multicast (PIM) [11], and Distance Vector Multicast Routing Protocol (DVMRP) [12]. Based on these multicast routing protocols, the streaming media are transported by Real Time Transfer Protocol (RTP) [13]. In the proposed multicast-based seamless streaming architecture, the delay, loss and service interruption caused by vertical handoffs or traffic congestion can be alleviated obviously due to the nature of duplications in multicast. Besides, to reduce the waste of bandwidth due to the duplicate traffic, an access network selection strategy that is capable of adapting to the dynamic traffic environment of the heterogeneous mobile networks is proposed. The proposed architecture is implemented in the UNIX environment, and five experiments are designed and executed to demonstrate the correctness and feasibility of the proposed architecture. The experimental results show that the video streaming service can be delivered smoothly during vertical handoffs. Moreover, it is found that the traffic overhead, including both of data and signaling packets, is only 1.0368% per vertical handoff in average for each mobile access network, and thus the feasibility of the proposed architecture is demonstrated.

The paper is organized as follows. Assumptions and system modeling are made in Section 2. The architecture of the proposed multicast-based multihoming for seamless streaming media services is described in Section 3. In Section 4, the proposed system is implemented in all-IPv6 environments. Five experiments are designed and the results are discussed. Finally, Section 5 gives the conclusions.

2 Assumptions and modeling

Assumptions of the heterogeneous mobile network environment are made first as follows. The receivers of the streaming media service are multihomed hosts with N_{Total} network interfaces attached to N_{Total} heterogeneous mobile access networks. The i th access network interface, NI_i , of the multihomed host interfaces with

the i th mobile access network, AN_i , where $1 \leq i \leq N_{Total}$. All mobile access networks form the set S_{AN} , where the size of S_{AN} , $|S_{AN}|$, is equal to N_{Total} . The multihomed mobile receiver and the network routers in all access networks S_{AN} are multicasting enabled. Associated with each network interface NI_i , a priority number P_i is given to reflect the preference of the mobile user, where $1 \leq i \leq N_{Total}$. The less the P_i , the higher the preference. Without loss of generality, let $P_i < P_j$ if $i < j$, where $1 \leq i, j \leq N_{Total}$. Let G_{MC} be the multicast group that is assigned to a specific receiver, and the address of the multicast group is $addr_{MC}$. The multihomed receiver can join G_{MC} in advance through one or more access network interfaces. These network interfaces form the set $S_{NI} = \{NI_i/NI_i \in G_{MC}, 1 \leq i \leq N_{Total}\}$. Multihomed mobile receivers join/leave a multicast group by using the Multicast Listener Discovery (MLD) protocol [14]. Multicast routers adopt the MLD protocol to find the mobile receivers who want to join/leave a multicast group, and to inform other multicast routers. The mobile access networks that the multihomed receiver is currently joining G_{MC} through the network interfaces S_{NI} form the multicast set $S_{MC} = \{AN_i/NI_i \in G_{MC}, 1 \leq i \leq N_{Total}\}$. The sizes of the sets S_{NI} and S_{MC} are the same, $|S_{MC}|$, where $|S_{MC}| \leq N_{Total}$. In general, the number of the access networks adopted for the delivery of the streaming media service is N_S , where $1 \leq N_S \leq N_{Total}$. During vertical handoffs, the number of the access networks adopted for the delivery of the streaming media service will be N_{VF} . Obviously, $N_{VF} \geq N_S$ for protection of the streaming media. The multicast set S_{MC} that will be adopted during the vertical handoffs is called \tilde{S}_{MC} , where $|\tilde{S}_{MC}| = N_{VF}$. In each access network, enough bandwidth and QoS strategies are supported to play the requested streaming media service smoothly. The quality of the i th access network AN_i is reflected by the parameter q_i , e.g., delay, packet loss rate and jitter. All q_i form the quality set $Q = \{q_i | \forall AN_i \in S_{AN}, 1 \leq i \leq N_{Total}\}$. Each q_i of AN_i is associated with a predefined threshold T_i . All the T_i form the threshold set $T = \{T_i | \forall AN_i \in S_{AN}, 1 \leq i \leq N_{Total}\}$. The RTP protocol is adopted to transport the streaming media.

3 Architecture of proposed multicast-based multihoming

Figure 1 shows the proposed multicast-based multihoming architecture that is capable of offering seamless streaming media services across heterogeneous mobile access networks. Initially, the receiver issues a request for a multimedia stream to the media server through NI_i . The source address of the request is set to

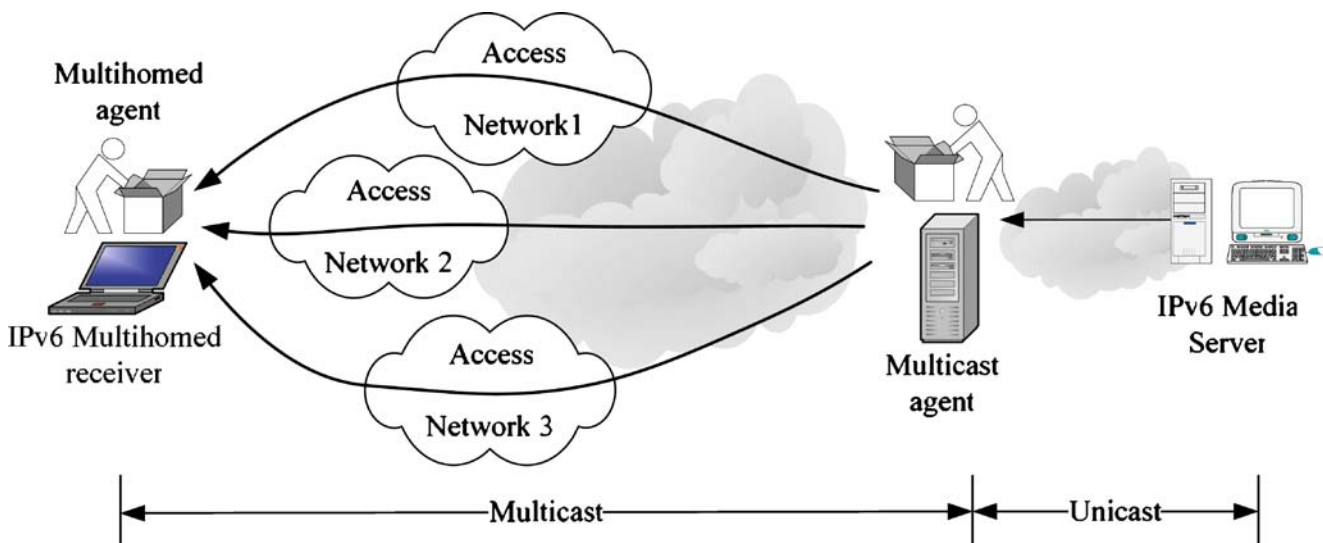


Figure 1 Architecture of the proposed seamless streaming media

be $addr_{MC}$. After receiving the request, the media server sends the requested streaming media to the multicast agent (MCA) by unicasting first. The multicast agent (MCA) then duplicates and forwards the streaming media packets, by multicast, to $addr_{MC}$ through all of the mobile access networks in S_{MC} . Thus the multihomed receiver will receive multiple copies of the streaming media packets from all $NI_i \in G_{MC}$. The multihomed agent (MHA) inside the receiver is responsible for the merging of the duplicate streaming media packets. The number of copies for the duplicate streaming media is equal to the sizes of the set S_{MC} , $|S_{MC}|$. Too many duplicate streaming media packets waste the network resources, although the network failures can be prevented, Therefore, which network interfaces, NI_i , and when they will be selected to join/leave G_{MC} for the receipt of the streaming media should be carefully determined, so that the overhead is reduced while the media stream is kept smooth and seamless.

Ideally, the bandwidth has no waste if $N_{VF} = N_S = |S_{MC}| = 1$. However, in the situation, the streaming media may be delayed or interrupt due to vertical handoffs or network congestion. In the proposed multicast-based multihoming architecture, even though an access network is unstable, unreachable or loss of signals, the receiver can still receive the streaming media on time from other access networks. Thus the more $|S_{MC}|$ is, the more the degree of the protection is. The value of $|S_{MC}|$ will be set to N_{VF} for protection, as the procedure for vertical handoff is activated. As the vertical handoff procedure finishes, set $|S_{MC}|$ to be N_S . Another mission of the MHA is to monitor the network status of all attached mobile

access networks. If the mobile access network that is currently adopted to deliver the streaming media is unstable or congested, the MHA will initiate the network selection procedure for vertical handoffs. Define that

$$\text{Access network } AN_i \text{ is stable if } q_i \text{ satisfies } T_i \quad (1)$$

It means that AN_i is stable if the constraint on the quality parameter q_i , say, $q_i \leq T_i$, holds. That is, the delay or the packet loss rate, q_i , is less than the predefined threshold T_i . If q_i does not satisfy T_i , the access network AN_i is treated to be unstable and the streaming media will be vertically handed over from AN_i to other stable access networks. Only the stable access networks will become the candidates for the selection of the target access networks to be joined for vertical handoffs. These candidates form the set $S_C = \{AN_i | AN_i \text{ is stable, } \forall AN_i \in S_{AN}, 1 \leq i \leq N_{Total}\}$. Without loss of generality, in the candidate set S_C , the access network AN_i is prior to AN_j if $i < j$. In the candidate set S_C , Q_C satisfies T_C , where $Q_C = \{q_j | AN_j \text{ is stable, } \forall AN_j \in S_C, 1 \leq j \leq |S_C|\}$ and $T_C = \{T_j | T_j \text{ is stable, } \forall AN_j \in S_C, 1 \leq j \leq |S_C|\}$.

The MHA measures and monitors the quality set Q for all access networks S_{AN} and then updates S_C with the period of T_P , even if there exists access networks in S_{AN} that are not reachable. Obviously, the set S_{MC} is a subset of S_C , and $|S_{MC}| \leq |S_C|$. As long as $|S_{MC}| \geq 1$, the multihomed receiver can keep the streaming media service alive. The MHA will join or rejoin G_{MC} through each access network in S_{MC} . Moreover, according to the measured quality set Q , the MHA can balance the traffic loads among all network interfaces of the multihomed host by using simple dispatching schemes. The

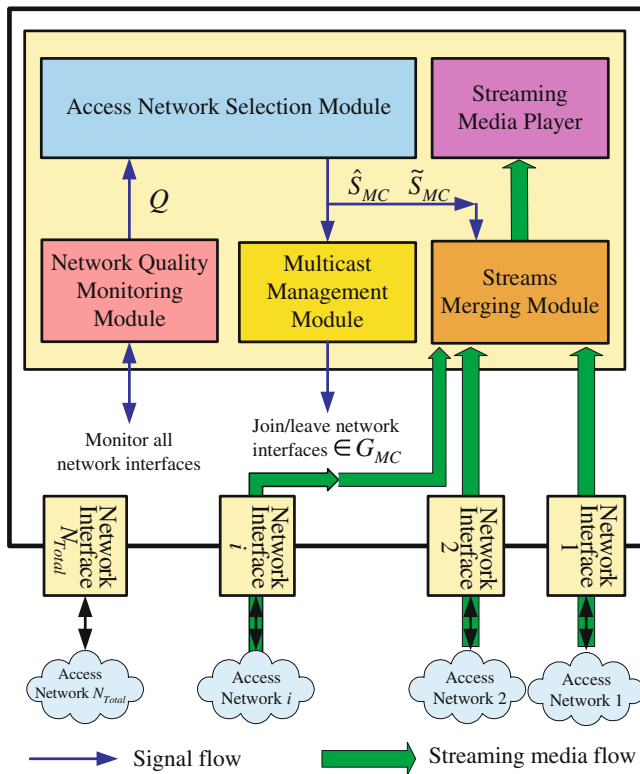


Figure 2 Functional structure of the multihomed agent for the proposed multicast-based multihoming architecture

strategy to select S_{MC} from the set S_C is performed by the multihomed mobile host and will be described later.

Figure 2 shows the functional structure of the MHA for the proposed multicast-based multihoming architecture. There are four modules in the multihomed mobile host: Network Quality Monitoring (NQM) module, Access Network Selection (ANS) module, Multicast Management (MM) module, and Streams Merging (SM) module. The NQM module measures and monitors the quality set Q for all network interfaces of the multihomed host periodically, and then sends the measured quality set to the ANS module.

The ANS module updates the candidate set S_C by comparing the received Q with T . Let the function

$\Psi(S, n)$ truncates the set S into a set with length n , i.e., the output of the function is a set that is composed of the first n elements in the set S . If $|S| < n$, the output of the function $\Psi(S, n)$ is just S , i.e., $\Psi(S, n) = S$ if $|S| < n$. The elements of S_C are arranged in ascending order of the priority number P_i , that is, the access network AN_i with less i has higher preference. Thus the set S_{MC} is obtained by truncating S_C .

$$S_{MC} = \Psi(S_C, N_S) \tag{2}$$

The set S_C updates its elements continuously, but the elements of the set S_{MC} will not be changed except that the vertical handoff procedure is triggered. Define that the system is in stable state only if all of the access networks $AN_i \in S_{MC}$ are stable. As long as the NQM module detects at least one of the access networks in S_{MC} is not stable, the system will leave the stable state at that moment, and enter the vertical handoff state. Let the set \hat{S}_{MC} be the access networks to be used for the delivery of the streaming media after the system leaves the vertical handoff state, where $|\hat{S}_{MC}| = N_S$. Whenever the system enters the vertical handoff state, two sets, \tilde{S}_{MC} and \hat{S}_{MC} , have to be determined first. The set \hat{S}_{MC} is composed of the first N_S elements in the newly updated S_C at the beginning of the vertical handoff procedure, i.e., $\hat{S}_{MC} = \Psi(S_C, N_S)$. The set \tilde{S}_{MC} is

$$\tilde{S}_{MC} = \Psi(S_{MC} \cup \hat{S}_{MC}, N_{VF}) \tag{3}$$

The two sets, \tilde{S}_{MC} and \hat{S}_{MC} , will be sent to both of the MM module and the SM module in the beginning and the end of the vertical handoffs, respectively.

The MM module is responsible for the execution of joining/leaving the IPv6 multicast group G_{MC} for vertical handoffs, according to both of \tilde{S}_{MC} and \hat{S}_{MC} informed by the NQM module. During the vertical handoff procedure, the MM module will not only keep the existing N_S connections over the access networks $AN_i \in S_{MC}$, but also establish additional $N_{VH} - N_S$ connections to receive the streaming media by joining the IPv6 multicast group G_{MC} through the network interfaces of the access networks $AN_j \in \tilde{S}_{MC} - S_{MC}$. After

Figure 3 State transition diagram

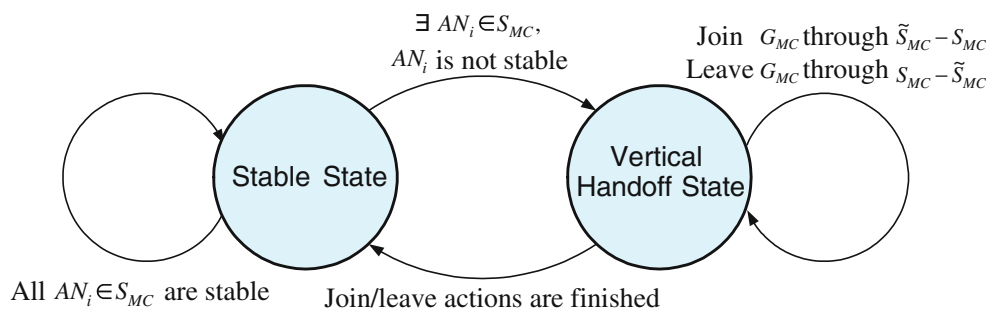
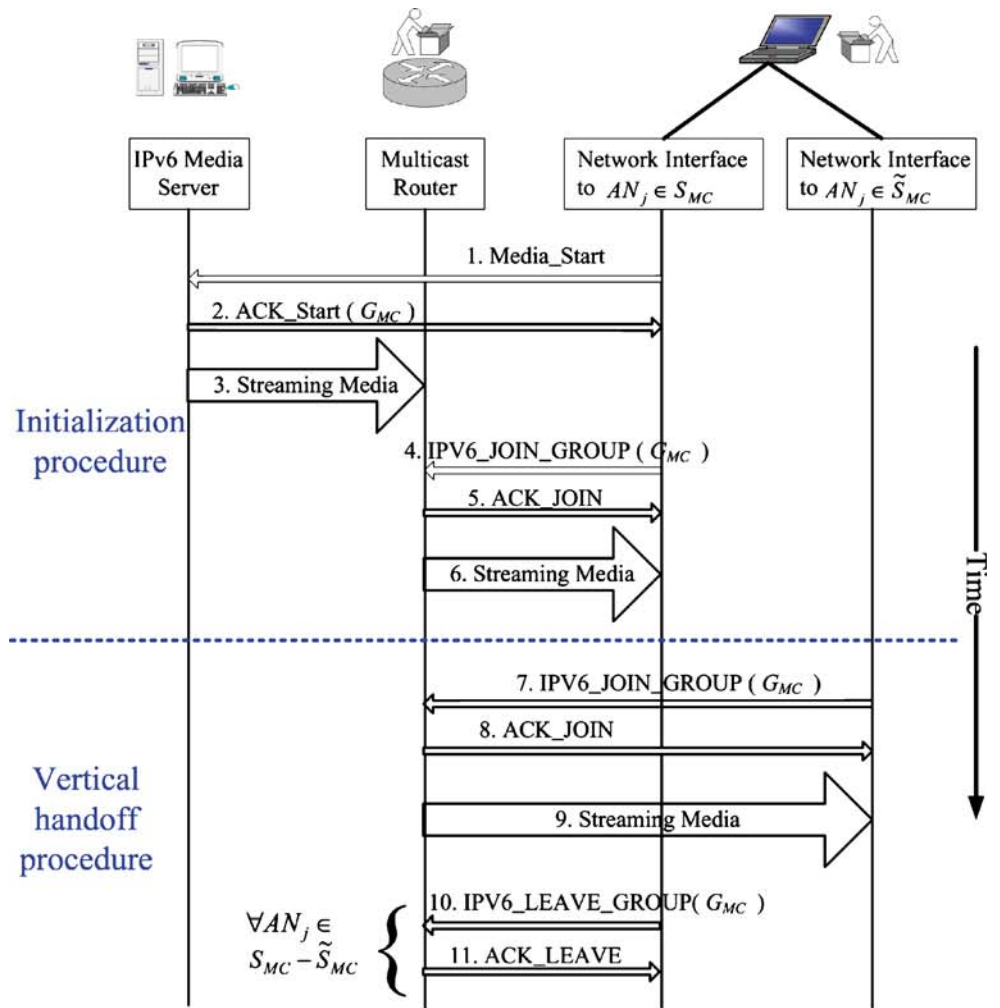
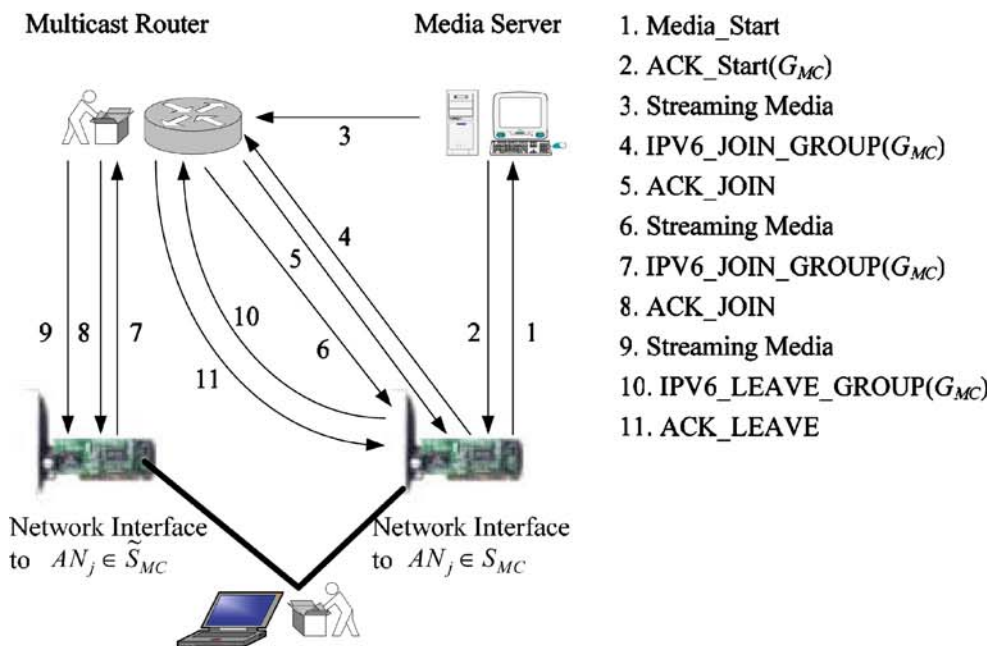


Figure 4 Signaling procedure operated in the MM module

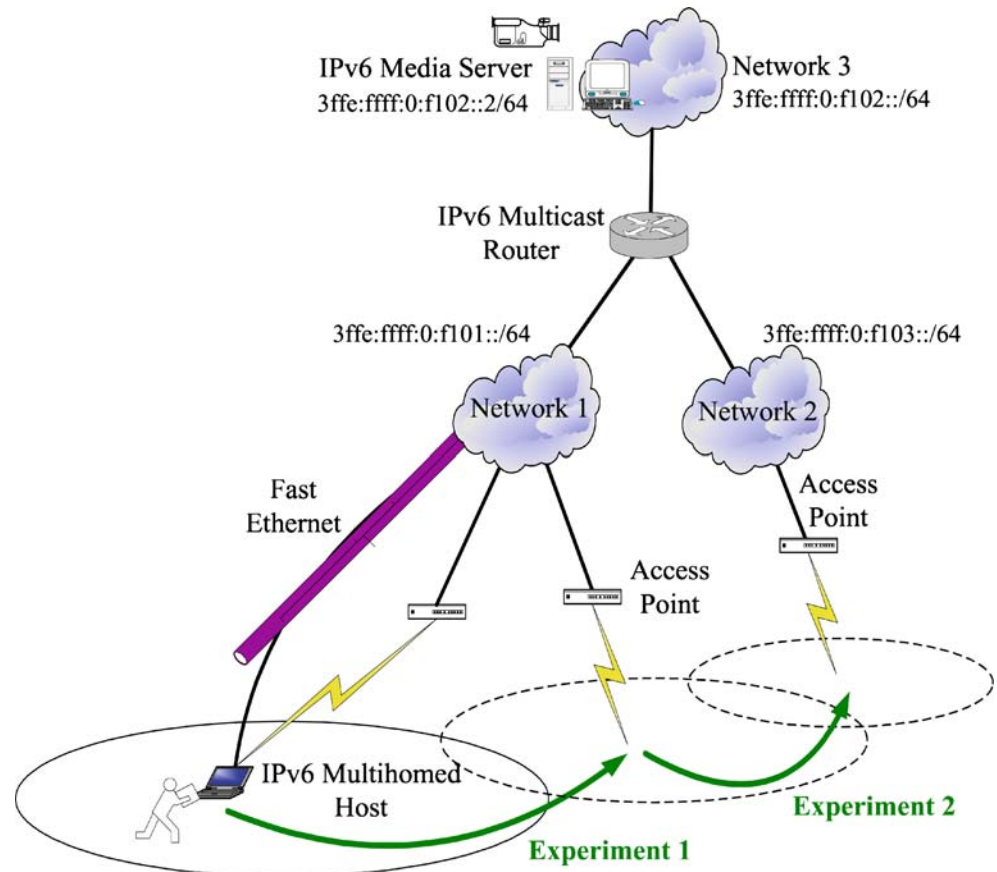


(a)



(b)

Figure 5 The network environment for experiments



the success of the establishment for the additional $N_{VH} - N_S$ connections, the MM module will leave G_{MC} through the network interfaces of the access networks $AN_j \in S_{MC} - \tilde{S}_{MC}$, so as to terminate the existing connections that pass through the access networks $AN_j \notin \tilde{S}_{MC}$. If all $AN_i \in S_{AN}$ are not stable, i.e., $S_C = \phi$ or $|S_C| = 0$, the set S_{MC} is an empty set also. Then the MM module will not perform the joining/leaving operations. As the above joining/leaving operations are finished, the system will leave the vertical handoff state, and return to the stable state. The state transition diagram is shown in Fig. 3. The signaling procedures of initialization and vertical handoffs in the MM module are shown in Fig. 4. In the initiation of the streaming service, the active network interface will send out the signal Media_Start to the media server. After the sending an ACK_Start signal back the media server will start multicasting by transmitting the RTP streaming media to the multicast router. The network interface will try to join the multicasting group by sending IPV6_JOIN_GROUP signal to the multicast router. The multicast router will acknowledge the join message by sending ACK_JOIN signal and start to send the RTP streaming media to the network

interface. When the signal of the active interface is measured to be low and a handoff is estimated to be happened, the system will select a new interface access to $AN_i \in \tilde{S}_{MC}$ to join the multicast group. The chosen interface will send IPV6_JOIN_GROUP signal to the multicast router and receive the acknowledgement and the streaming media hereafter. In order not to waste the resource of the network, the previous network will leave the multicast group after the new network interface start to receiving streaming media, no matter the handoff of the previous network interface taken place or not.

The SM module is responsible for the merging the duplicate $|S_{MC}|$ media streams received from the network interfaces currently joined in the IPv6 multicast group G_{MC} . To identify the redundant packets, the SM module will record and check the sequence number of every received RTP packet. The RTP packets that have ever seen will be dropped. Only one of the $|S_{MC}|$ duplicate RTP packets will be forwarded to the streaming media player.

The heterogeneous access network technology that users can access to nowadays includes 3G, Wi-Fi, WiMAX and so on. Each of them has its unique

Table 1 Development environment of the implemented system

System part	Name and version
IPv6 multihomed host	
Notebook	Acer 341T
OS	Debian GNU/Linux, testing distribution kernel-2.4.19
Wireless LAN	Orinoco 802.11b card
IEEE 802.11b access point	Lucent RG-1000
Wired LAN	Intel eeepro100 card
VideoLAN-VLC	vlc-0.6.2
IPv6 media server	
OS	Debian GNU/Linux, testing distribution kernel-2.4.19
VideoLAN-VLC	vlc-0.6.2
IPv6 multicast router	
OS	FreeBSD 4.6.2-release
Multicast routing daemon	Pim6sd, RP (Rendezvous Point)
IPv6 autoconfiguration daemon	Radvd-0.7.2-1

characteristics, but all of them try to provide streaming media services. 3G, which provides the least bandwidth of them, promise sufficient bandwidth up to 384 kbps to support streaming media services. Not to mention the Wi-Fi and the WiMAX that provides 54 Mbps and 100 Mbps, respectively. Besides, there are many researches about the providing of streaming media services on these access network technologies [15–18]. But the variety of different access network is still the biggest obstacle for providing streaming services with all the access networks together. Solutions aimed at the MAC layer are difficult to cover all the networks. The design of the proposed architecture only depends on the support of multicasting of the networks, and is independent on different access network technologies.

That means as long as the access network technologies support multicasting, the proposed architecture will work.

4 Implementation and experimental results

The proposed multicast-based IPv6 multihoming system is implemented in the IPv6 network environment shown in Fig. 5, where the VLC media player developed by the VideoLAN [19] project is modified and extended here. There are an IPv6 media server, an IPv6 multihomed host, and an IPv6 multicast router in the implemented system. The IPv6 multicast router has three IPv6 subnets which are 3ffe:ffff:0:f101::/64, 3ffe:ffff:0:f102::/64, and 3ffe:ffff:0:f103::/64. The address of the IPv6 multicast group G_{MC} , $addr_{MC}$, is ff08::1. For the lack of the assistance from GPRS carriers, the Fast Ethernet network is adopted as the data transmission medium for GPRS, and the strength of the GPRS signals is simulated by a program executed in the IPv6 multihomed host. IEEE 802.11b WLAN is another access network. The IPv6 multihomed host is equipped two network interface cards to Fast Ethernet and 802.11b WLAN networks, and is able to receive streaming media from both of the two access networks simultaneously. Thus $|S_{AN}|=N_{Total} = 2$. For the Ethernet interface provides higher bandwidth than the WLAN interface, a higher preference, or a smaller priority number, is given to the Ethernet interface. Let AN_1 and AN_2 represent the Ethernet and WLAN networks, respectively, $S_{AN}=\{AN_1, AN_2\}$. The names of the network interfaces are called eth0 for Ethernet and eth1 for WLAN in the IPv6 multihomed host. Let $P_1=1, P_1=2, N_S=1, N_{VF}=2, T_p=1, T_1=q_2,$ and $T_2=q_1$, where q_i is the link quality. The MHA and the MCA is implemented in the IPv6 multihomed host and the IPv6 multicast router, respectively. The VLC server

Table 2 Tools, libraries and their version adopted in the development of MHA

System part	Name and version
IPv6 multihomed host	
C compiler	gcc compiler.gcc-3.3.1
Packet capture	Ethereal-0.9.13-1
VideoLAN-VLC	vlc-0.6.2
Related library	ffmpeg-20030813 libdvbpsi-0.1.3 libmad-0.15.0b mpeg2dec-20030612 xlibs-dev-4.2.1-6

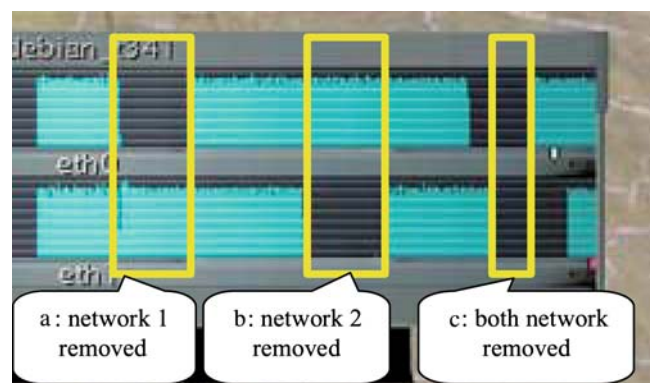


Figure 6 Historical RTP packet statistics of both access networks

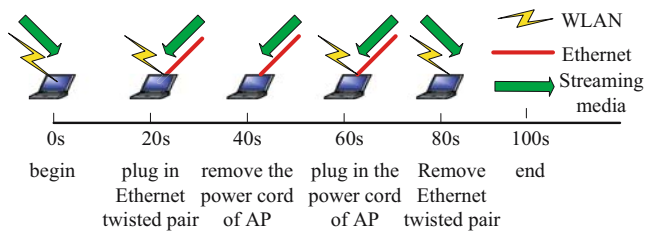


Figure 7 Experimental scenario of network failures

and the modified VLC client are installed in the media server and the IPv6 multihomed host individually. The format of the videos provided by media server is MPEG 1.

There are three PCs play the roles of IPv6 multihomed host, IPv6 multicast router and IPv6 media server separately. IPv6 multihomed host has an Intel Pentium III 450 MHz CPU, 192 MB memory and a 5.4 GB Hard disk. IPv6 multicast router equipped with an Intel Pentium III 450 MHz processor, 256 MB memory and a 10 GB hard drive. And the specification of the IPv6 media server is an Intel Pentium III 450 MHz CPU, 289 MB memory and a 6 GB hard disk. Table 1 lists the development environment adopted in the system. In the IPv6 multicast router, the pim6dd daemon are installed for multicast routing, and the radvd daemon, router advertisement daemon for IPv6, is also installed for address auto configuration. The structure of ipv6_mreq and the function of setsockopt are used to join/leave the IPv6 multicast group. Table 2 lists the tools, libraries and their version adopted in the development of the

MHA in the IPv6 multihomed host. The tool, ethereal, is adopted to capture packets. The following libraries are required for compiling and installing VLC in the IPv6 multihomed host. The ffmpeg library is a free MPEG-4/DivX/OpenDivX codec software. The libdvbpsi library is the decoder for processing streaming media in VLC. The libmad library is the mp3 decoder, and mpeg2dec is the decoder for MPEG1 and MPEG2. The library, xlibs-dev, is used for image output of basic x11.

Four experiments are designed and executed to verify the functionality and the performance of the proposed system.

4.1 Manual vertical handoffs enforced in the same IPv6 subnet

Both of Ethernet and wireless LAN interfaces of the IPv6 multihomed receiver are connected to the same subnet, Network 1 (3ffe:ffff:0:f101::/64). Network failures are used to enforce the handoffs between Ethernet and wireless LAN, and it is emulated by removing manually the twisted pair from the IPv6 multihomed host or the power cord from the access point. Figure 6 shows the historical RTP packet statistics of both access networks. The twisted pair is removed from the IPv6 multihomed host in rectangle A, and the power cord is removed from the access point in rectangle B. In both cases of network failures, the streaming media service is still continued by receiving media packets

Figure 8 Sequence number of RTP packets and their arrival time in Experiment 1

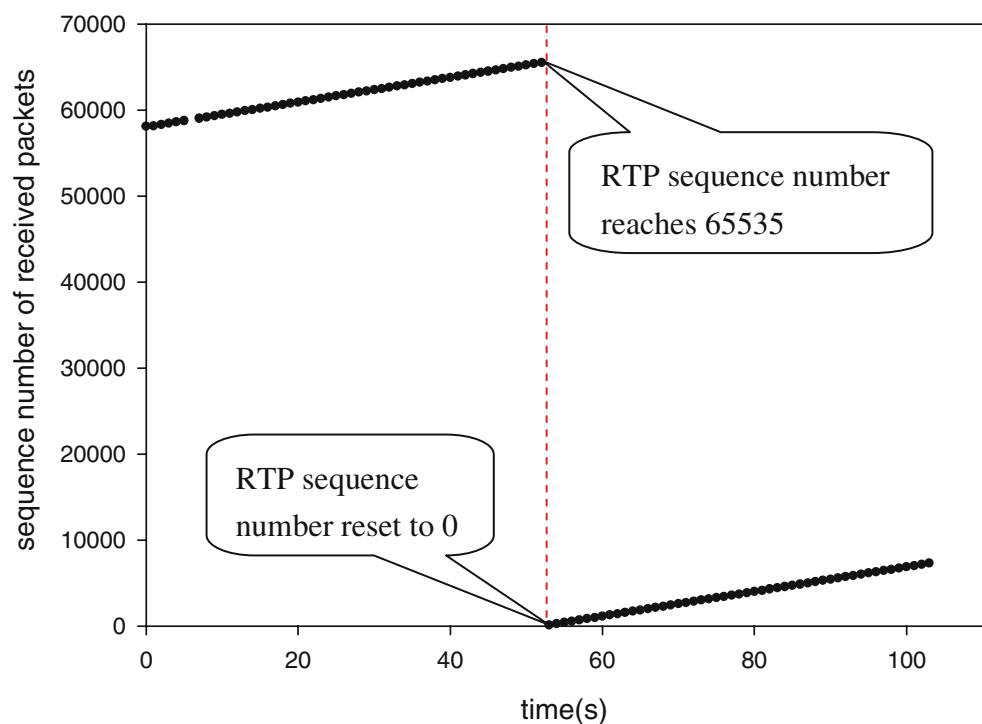
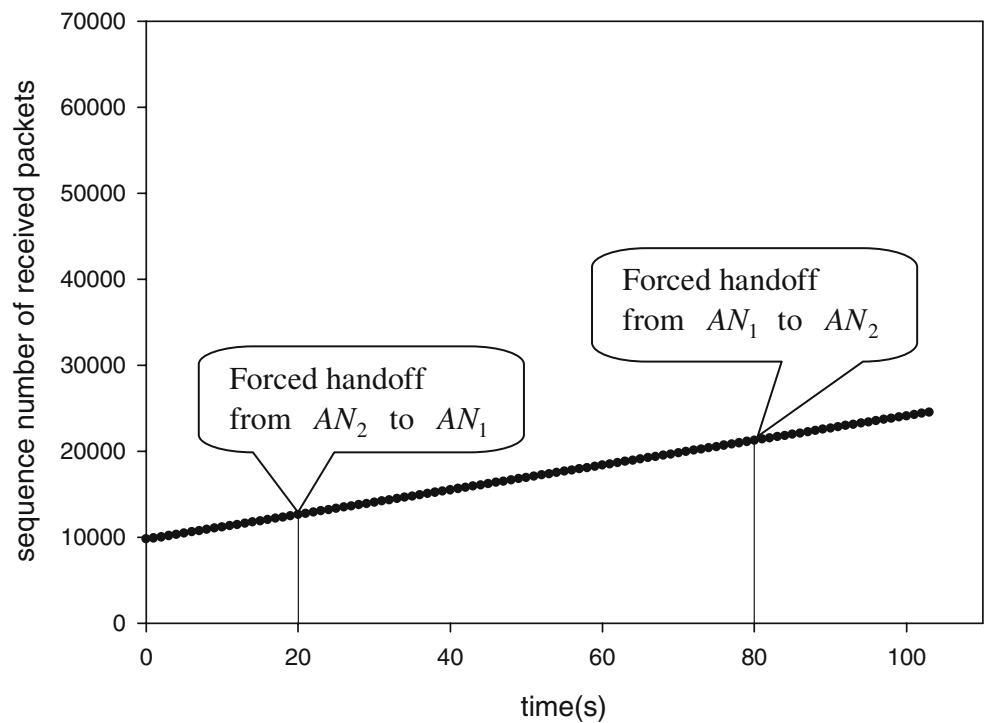


Figure 9 Sequence number of RTP packets and their arrival time in Experiment 2



from the other network interface. It is observed that the streaming media service will be interrupted only if both networks are failed, as indicated in rectangle C.

The scenario of network failures is shown in Fig. 7 to force the vertical handoffs: (1) initially, at $t=0$, only WLAN is connected, (2) at $t=20$ s, the Ethernet twisted pair is plugged in, (3) at $t=40$ s, the power cord of the access point is removed, (4) at $t=60$ s, the power cord of the access point is plugged in, and (5) at $t=80$ sec, the Ethernet twisted pair is removed. The network interface that should deliver the streaming media mainly is also shown in the figure. The experiment demonstrates that the operations of the multihomed host are correct, and the multicast set S_{MC} is

$$S_{MC} = \begin{cases} \{AN_1\}, & 20 < t \leq 80 \\ \{AN_2\}, & 0 < t \leq 20 \text{ and } t \geq 80 \end{cases} \quad (4)$$

Figure 8 shows the sequence number of the received RTP packet versus their arriving time under the scenario. The figure shows that, as the upper bound that RTP can process, 65,535, is reached, the sequence number is reset to zero and accumulated again. It demonstrates the success for the operations of the SM module. Moreover, The figure shows all RTP packets can be delivered, without any loss, to the streaming media player in order and on time. Therefore, the seamless handoff is achieved.

4.2 Manual vertical handoffs enforced across different IPv6 subnets

In the experiment the multihomed receiver connects to both access networks with different IPv6 subnets, Network 1 (3ffe:ffff:0:f101::/64) for Ethernet and Network 2 (3ffe:ffff:0:f103::/64) for WLAN. Figure 9 shows the sequence number of the received RTP packet versus their arriving time under the scenario given in Fig. 7. The vertical handoffs occur at $t=20$ s and $t=80$ s, and all RTP packets are continuously delivered, without any loss, to the streaming media player in order and on time. That is, the quality of the streaming media is not affected. Therefore, the seamless vertical handoffs are achieved even if the two network interfaces are connected to different subnets.

For the basic experiments, the implemented system provides smooth visual results during handoffs, and below we analyze the situations that will break the smooth stream playing in the proposed architecture. In the proposed architecture, Let B be the probability that the smooth stream playing will be interrupted, then:

$$B = \sum_{i=1}^{N_{Total}} A(N_i) \prod_{i=1}^{N_{Total}} L(N_i) \quad (5)$$

Where $L(N_i)$ denotes the probability that the signal of network interface N_i is low, and $A(N_i)$ is the probabil-

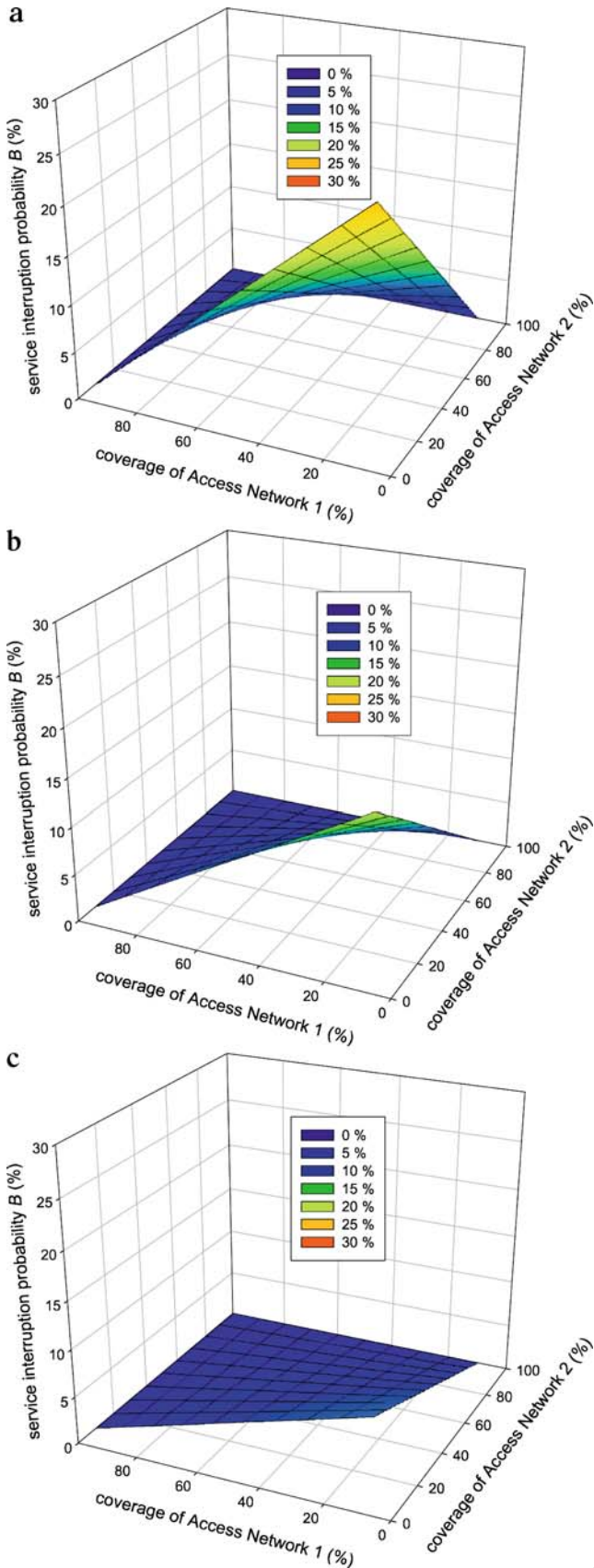


Figure 11 The effects of congested WLAN on the streaming media.

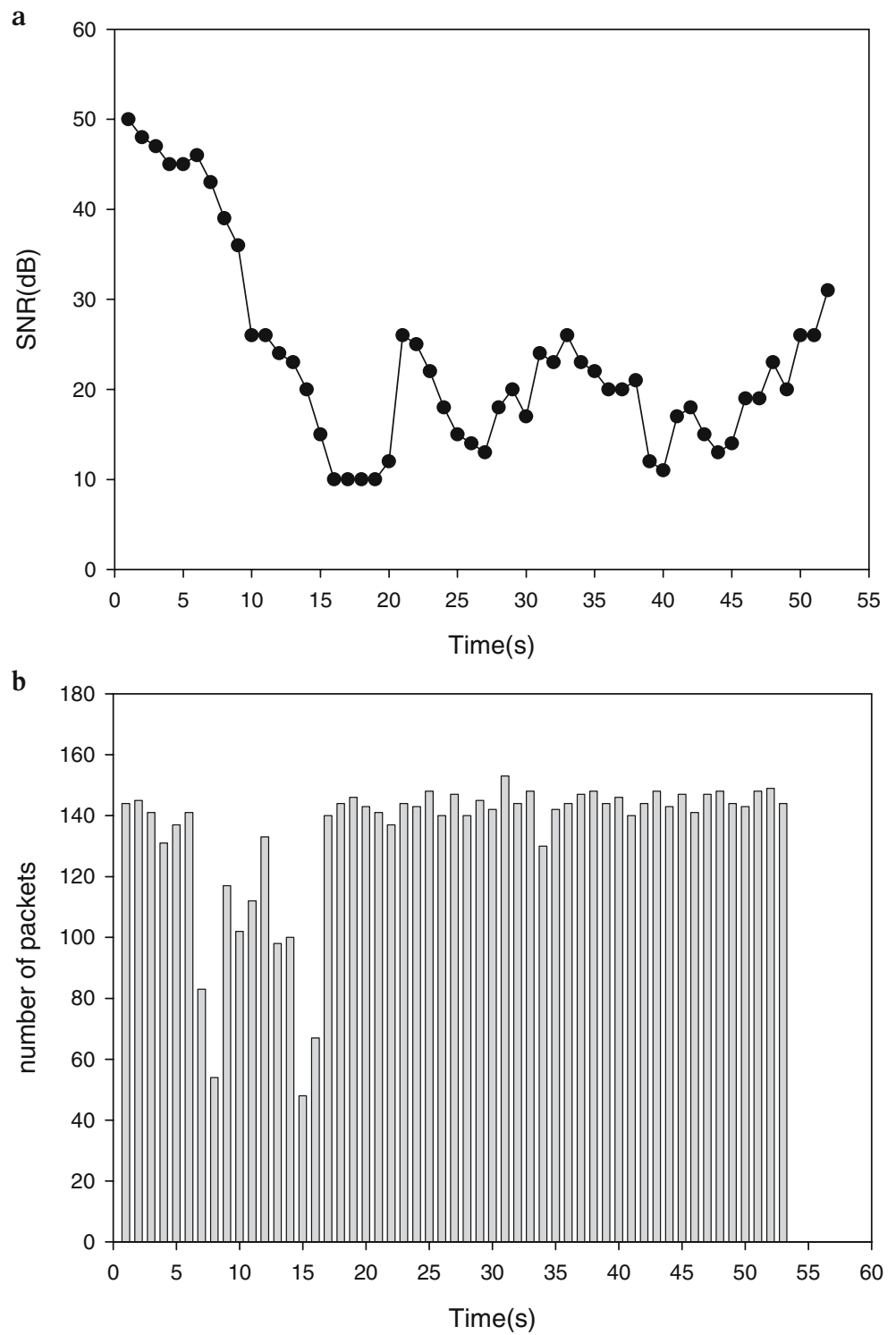
ity that network interface N_i is active for receiving streaming. Consider a heterogeneous network environment that consists of three different kind of access networks, and each of them has a 70% coverage. Under such condition, $\sum_{i=1}^{N_{Total}} A(N_i) = 1$, $\prod_{i=1}^{N_{Total}} L(N_i) = (1 - 0.7)^3 = 0.027$, and streaming broken rate will then be: $B = 1 \times 0.027 = 0.027$. The relationship between coverage and stream interrupted probability is shown in Fig. 10.

4.3 Effect of congested WLAN and vertical handoffs on streaming media

Initially, the multihomed host connects to the WLAN only, i.e., $S_c = S_{MC} = \{AN_2\}$. By moving the multihomed host away from the access point gradually, the WLAN is then getting unstable, i.e., q_2 does not satisfy T_2 . The situation can be treated as network congestion. Figure 11 shows the link quality of WLAN, the number of RTP packets received from WLAN, and the displayed streaming video. It is found that the RTP packets will be lost if $q_2 \leq 10$, where $T_2 = 10$. Therefore, the discontinuity occurs in the display of the streaming video. Figure 12(a) and (b) show the Signal-to-Noise Ratio (SNR) and the number of the received RTP packets versus time, respectively. As the multihomed host moves away from the access point, the detected SNR is reduced to 15–20 dB. It is found that, starting from $t = 7$ s, the RTP packets suffer a large amount of loss, and thus the wireless connection becomes unstable. At $t = 17$ s, the Ethernet twisted pair is manually plugged in. Thus a request to join the multicast group G_{MC} from

◀ **Figure 10** Coverage of access networks versus stream interrupted probability. (a) coverage of access network 3 is 70% (b) coverage of access 3 is 80% (c) coverage of access network 3 is 90%.

Figure 12 The effect of vertical handoffs on the streaming media.(a) SNR versus time (b) the number of merged packets versus time.



NI_1 , and another request to leave G_{MC} from NI_2 are issued. Henceforward the merged RTP flow is still smooth, no matter how the SNR changes. It means that the streaming media service is vertically and smoothly handed over from WLAN to Ethernet networks, as WLAN is congested.

4.4 Automatic vertical handoffs with simulated link quality for GPRS

As described in the beginning of the section, the GPRS or 3G networks, instead of Ethernet, should be adopted in the real world. However, the assistance from Tele-

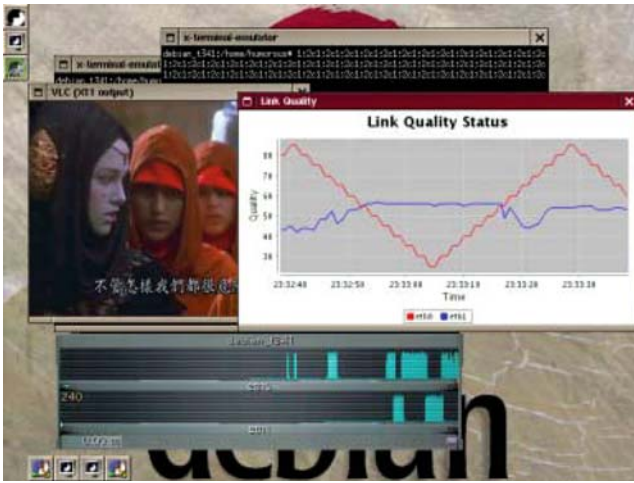


Figure 13 Automatic vertical handoffs by using the real link quality measured in WLAN and simulated link quality for GPRS (Ethernet).

communications carriers is lacked. Thus Ethernet is used to be treated as the transmission medium of GPRS, and its signal strength or link quality is given by a mathematic function $f(t)$ to simulate the roaming situation for a mobile user, where

$$f(t) = \begin{cases} 100 - \left(\left[\frac{t}{100}\right] * 100 - t\right) & \text{if } \left[\frac{t}{100}\right] \text{ is even} \\ \left[\frac{t}{100}\right] * 100 - t & \text{if } \left[\frac{t}{100}\right] \text{ is odd} \end{cases} \quad (6)$$

Figure 14 User speed versus overhead.

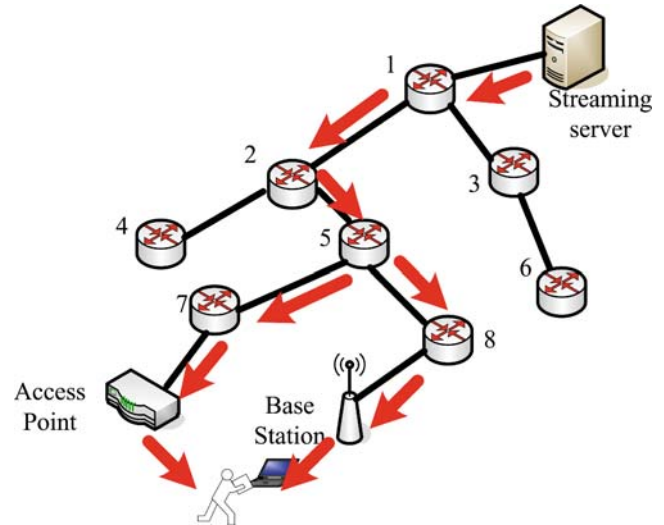
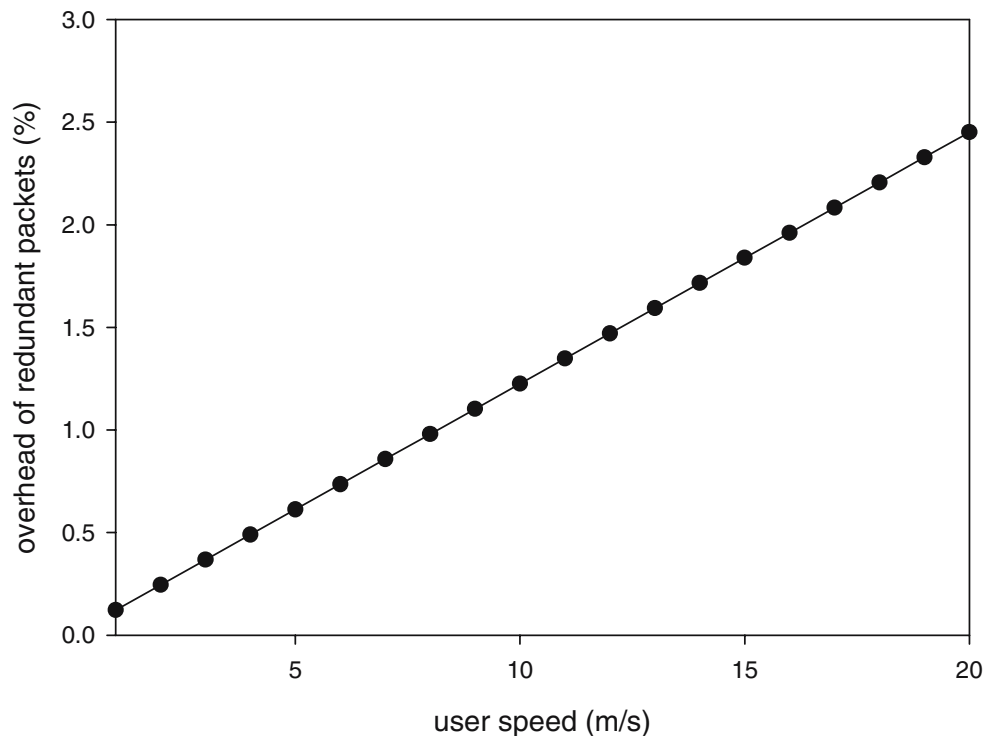


Figure 15 Overhead for multicasting stream media.

As shown in Fig. 13, the red line and the blue line indicate the simulated link quality of GPRS and the real link quality measured from WLAN. Vertical handoffs are thus performed automatically, so that the functions and operations of the four modules, NQM, ANS, MM, and SM, in the MHA can be checked. The experimental results demonstrate that all of the four modules operate correctly so as to achieve the seamless vertical handoffs.

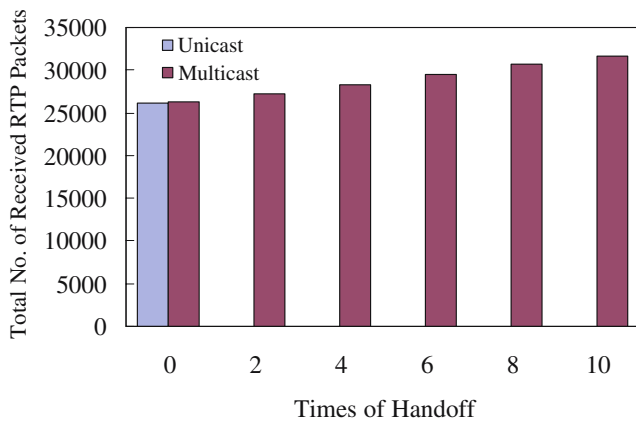


Figure 16 Total number of RTP packets received from both networks.

4.5 Overhead of vertical handoffs

The major overhead of the proposed architecture is the redundant packets sent to the receiver via multiple interfaces during vertical handoffs. Let total packets received by the receiver t be p , and the redundant packets received during a vertical handoff be p_r , the overhead of the redundant packet is then $\frac{hp_r}{p}$, where h is the vertical handoff occurrence. If all the vertical handoffs succeed, the handoff h can be viewed as the occurrence of losing signal of the active network interface and can be defined as follows:

$$h = \sum_{i=1}^{N_{total}} L(N_i)A(N_i) \tag{7}$$

Take the heterogeneous network environment of 802.11 and 802.16 for example. For the pedestrian, the walking speed is about 1.2 m/s, and the coverage of 802.11 and 802.16 networks are 100 m and 6,000 km. Assume the APs and the BSs are well deployed and all vertical handoffs succeed, the probability that 802.11 and 802.16 networks are active are $\frac{100}{6,000+100}$ and $\frac{6,000}{6,000+100}$, respectively. And the handoff occurrence of the example environment h will then be:

$$h = \frac{1.2}{100} \frac{100}{6,100} + \frac{1.2}{6,000} \frac{6,000}{6,100} = 0.0004 \tag{8}$$

And for the vehicles that has an average speed of 12 m/s, in such network environments, the handoff occurrence h will be 0.004 per second. By measuring the system implemented, we found that the packets received per second is 145 in average and the packet received during each vertical handoff is 542 in average,

the overhead of the example network environment is then derived,

$$\frac{hp_r}{p} = \frac{0.0004 \cdot 542}{145} = 0.0015$$

The relationship between user speed and overhead is shown in Fig. 14. Because the proposed architecture multicasts the stream media, the overhead can be further reduced than simply duplicating streams. As shown in Fig. 15, if all routers are multicasting enabled, the stream will be duplicated below router 5. Compare with two individual streams, the hops between the streaming server and router 5 will not sending redundant packets.

Figure 16 shows the total number of RTP packets received from both of the WLAN and Ethernet networks, including the streaming media packets and all signaling packets, for the cases of unicasting and multicasting, where a video is delivered during a period of 180 s. For the cases of multicasting, the times of vertical handoffs, 0, 2, 4, 6, 8 and 10, are enforced during the 180 s period by adjusting appropriately the function of the simulated link quality for GPRS. It is found that the total number of the RTP packets received from both networks for the case of multicasting with zero vertical handoff times, 26,134 packets, approximates to that for the case of unicasting, 26,246 packets. Moreover, the total number of the RTP packets received from both networks is a linear function of the times of vertical handoffs, and the slope is 541.93 RTP packets per vertical handoff. The traffic overhead of each vertical handoff for the proposed multicast-based IPv6 multihoming architecture is $541.93/26,134=2.0737\%$ only. Thus the traffic overhead per vertical handoff in each access network for the proposed architecture is 1.0368% in average.

5 Conclusions and future works

Emerging wireless access network technologies brings the possibility of ubiquitous access to the Internet, but they also bring the variety. Heterogeneous access network environment will be a challenge for providing multimedia services seamlessly to the users roaming around. The multicast-based multihoming architecture is proposed and implemented in the paper to provide seamless streaming media services in the heterogeneous network environment. In the architecture, an access network selection strategy based on the measured network status is proposed. Five experiments are designed to verify the correction and the feasibility of the proposed multicast-based multihoming architecture. The access networks adopted as heterogeneous networks

in the experiments are WLAN and Ethernet networks, where the Ethernet network with a simulated quality value is adopted to emulate the GPRS network. The experiments demonstrate that the seamless streaming media service can be achieved in the proposed architecture during vertical handoffs. And the seamless streaming media service can also be achieved under network congestion. The coverage of each access network will affect the service interruption probability, and the relationship between coverage and the interruption probability is analyzed and illustrated. Overhead of redundant packets is also studied in the paper. It is found that the user speed is proportional to the overhead due to the frequent handoffs. The proposed multicast-based multihoming architecture is also shown to have less overhead on redundant packet than simply duplicate streams, since the traffic can be aggregated before the multicast router in the proposed architecture. The proposed architecture will introduce additional traffic in the heterogeneous mobile networks. In the experiments, it is found that the average traffic overhead per vertical handoff is 1.0368% only in each access network. And the analysis also shows that traffic overhead is 0.15% for the vehicles and 0.04% for the pedestrian.

The future works of the proposed architecture will focus on is discussed as follows. Session initiation protocol (SIP) is the standard for initiating, modifying, and terminating sessions of video, voice and instant messaging. The proposed architecture can be further combined with SIP for the session management and the support of personal mobility. There are several ways to combine SIP with the proposed architecture. First, the proposed architecture can use SIP only for the session management. The multicasting address negotiation process should be added to the SIP protocol to handle the user request. Second, multicast agent in the proposed architecture can be combined with SIP proxy, thus the multicast agent can be put into the deep of the network, but not the boundary of the access network. This increases the length of the unicast part of the route and reduces the redundant packets transmitted. At last there are some factors that user may concern such as power consumption or cost of the network. When selecting access networks, these factors can be taken into consideration.

Acknowledgments The authors would like to deeply thank the anonymous reviews' constructive suggestions and thoughtful comments that greatly improve the quality of the paper. This research was supported in part by National Science Council of the Republic of China under grant NSC 94-2213-E-008-018 and NSC 94-2219-E-260-006, and by Ministry of Economic Affairs under grant 95-EC-17-A-02-S1-029.

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