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Timed PR-SCTP for Fast Voice/Video over IP in Wired/Wireless Environments

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Abstract. *IIn this paper, we have successfully implemented a seamless handoff between heterogeneous networks to achieve a fast voice/video over IP (V^2oIP) with Partially Reliable Stream Control Transport Protocol (PR-SCTP) instead of TCP or UDP. In order to improve real-time video/voice streaming over IP, the conventional framework IHU has been remodeled as a new framework called SCTP-IHU that can simultaneously resolve the crucial problems of head-of-line blocking, handoff interruption, and non-real-time transmission using multi-streaming, multi-homing, and partial reliability in PR-SCTP. Four types of handoff experiment, switching paths (a) from wired to wireless network, (b) from wireless to wired network, (c) between wired networks, and (d) between wireless networks, have been tested successfully for real-time V^2oIP within the smallest delay (1.5 sec) for any handoff, that is, so-called timed PR-SCTP. As a result, the scheme SCTP-IHU definitely realizes real-time streaming multimedia for voice/video over IP (V^2oIP) in parallel. With high throughput, low handoff delay, and minimum packet losses rate, our proposed approach outperforms the previous work TCP-based USHA scheme.*

Keywords: PR-SCTP, SCTP-IHU, Multi-straming, Multi-homing, Partial reliability, V^2oIP

1. **Introduction.** In recent years, mobile wireless communication has also been considered and applicable to inter-platform mobile wireless communication between the embedded systems for which wireless communication has been composed by complex subnets. For example, vehicle-to-vehicle (V2V) communication is nowadays one of the most popular topics related to mobile wireless communication. The transmission of data between two entities utilizing IP is currently possible over fixed telephone and data networks. The evolution of mobile data networks has extended this capability to allow a communication link to exist within a cellular system. As the user moves from cell to cell, the data link is maintained through handoff techniques that are similar to those used to maintain a voice call. Particularly, it seems very interesting for many applications that streaming data services have been developed in hand-held device or in-vehicle set-top box operated in mobile wireless communications to access messages through WiFi, WiMax, or 3G wireless networks. A seamless handoff goal is to provide continuous connection for an end-to-end data transmission in the face of any link outage or handoff event. The two most design issues are low packet loss and low latency. Seamless handoff issue has been studied widely [10, 3, 6, 5]. The solutions fall into two categories: mobile IP [12] solutions and mobile IP-less solutions. The mobile IP solutions are typically based on Mobile IPv4 [13] or IPv6 [2] standards, requiring the deployment of different agents on the Internet for relaying and/or redirecting the data to the moving mobile host (MH). Most Mobile IP-less solutions implement a new session layer above the transport layer to make the application sessions transparent to the connection changes in the underlying layers [4, 15, 8, 14]. Other mobile IP-less solutions suggest new transport layer protocols such as SCTP [17] and TCP-MH [9] to provide the necessary handoff support. Transmitting network has to cooperate with each protocol layer. At present, network layer, transport layer, or application layer has the protocol to support the mobility management in the wireless network. Table 1 shows the each layer which can support the mobility management.

Table 1. Protocols support mobile node

Layer	Protocol
Application Layer	SIP [16]
Transfer Layer	M-SCTP [17]
Network Layer	MIPv4, MIPv6 [2, 13]

Providing multimedia applications such as video conferencing, real time audio and video streaming which are common nowadays in the Internet [18, 7, 11], is considered to be very challenging and demanding over wireless networks. In wireless networks limited bandwidth support and lack of Quality of Service (QoS) at the air interface restrain wide deployment of these multimedia applications. One of the factors that affect the QoS is the transport protocol. The application performance can vary significantly when different transport protocols are employed. Currently UDP is mostly used for the transport of multimedia applications. However, UDP performance is not satisfactory due to several constraints such as lack of congestion control mechanism and packet loss. Likewise, TCP have to recover the drop of packets by retransmission at the cost of severe time delay, even though it is reliable. Instead we focus on the alternative SCTP for streaming multimedia. Our objective in this study is to realize fast voice/video over IP (V^2oIP) across two Linux kernel based embedded platforms XSCALE_NAV270 [19] where the features of PR-SCTP [17] "multi-homing", "multi-streaming", and "partial reliability", as shown in Fig. 1 and Fig. 2, are activated to overcome the crucial problems of head-of-line blocking, handoff interruption, and non-real-time transmission. A graphic user interface

(GUI) based on Qt [21] shows a user-friendly control panel for monitoring link status and switching primary link to another one. Furthermore, we introduce SCTP-IP handler to detect any possibility of path failure during the transmission. In order to perform streaming multimedia across two embedded platforms effectively, the packet format of IHU [22] upon the control, audio, and video packets have been revised and redefined to achieve the success of real-time streaming multimedia between nodes and we refer it to as the framework SCTP-IHU. Here we take into account the issue of latency during a handoff. We do several trials and found out the minimum delay that can trigger a handoff completely. After that, we set the smallest delay as a threshold for any handoff when switching the primary path to the secondary. This way gives timed reliability services and thus it is called timed PR-SCTP. Four types of handoff experiment have been done successfully in this study that executed V^2oIP and tested the path switching (a) from wired to wireless network, (b) from wireless to wired network, (c) between wired networks, and (d) between wireless networks.

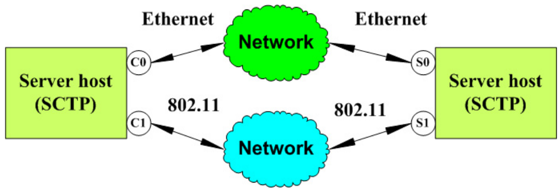


Figure 1. Multi-homing of a SCTP association

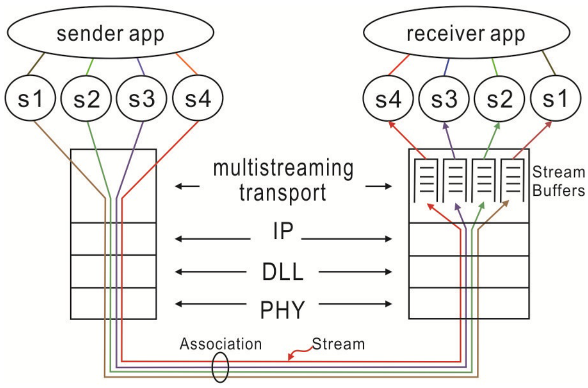


Figure 2. Relationship of an SCTP association to streams

2. **Path monitoring and selection.** In this paper, we propose a method to monitor the paths by using the SCTP socket API. How do we detect the path failure and when do we switch to the redundancy path? As shown in Fig. 3, we design a structure to record some information used to detect the status of paths. An object called Confirm Timer is designed for generating a timer interrupt to check the current situation about the connectivity in the primary path by every 200 milliseconds periodically. If the primary path fails to response any data receiving from the peer side within a specified duration, system will look for the alternative IP path to hand over from the original one. However, if alternative path is not available at the moment, system sends a warn message to users and then abort this association later on. When the host receives the data from a particular IP address, the system will update the receiving time of that particular IP address. If the host does not receive any data within a certain period, the system will declare that

particular IP inactive. After declaring the primary address inactive, the application will search an available IP from IP list to be primary address. If there is no available IP in the IP list, the application will warn the user and disconnect the communication. The flowchart shows in Fig. 4. For the primary path, the connectivity is detected by receiving data within a certain period of time. If the user at the remote host does not send any data to the local host, the system will send the confirm data (packet) periodically to let the local host know the remote IP is connected. For the other redundant paths, we use the other feature of SCTP, "heartbeat mechanism", the system will send HEARTBEAT chunk to the idle paths. The application can control the heartbeat and set the error threshold for an address. The error threshold is the number of missed heartbeats or retransmission timeouts that must occur before a destination address is considered unreachable. When the destination address becomes reachable, detected by heartbeats, the address becomes active.

```
class IPStruct
{
   public:
   // record the latest receiving time
   QTime recv;
   // record the latest sending time
   QTime send;
   // indicate the connectivity
   bool connectivity;
   // the confirm timer used to check the status of connectivity
   QTimer Confirm Timer;
};
```

FIGURE 3. Manager information structure of each peer IP

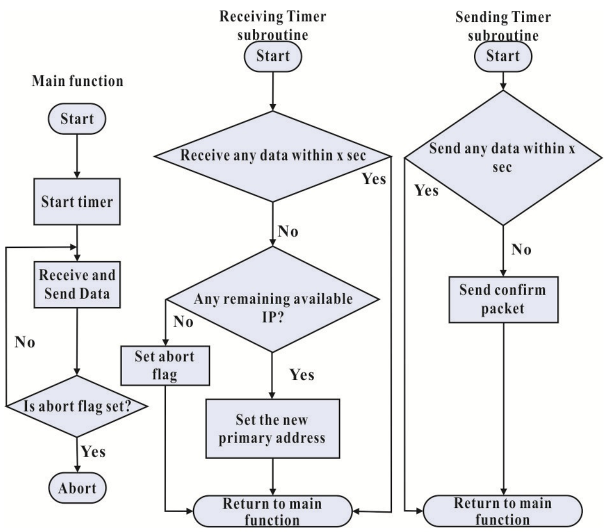


FIGURE 4. Flowchart of detecting timer

3. **System development.** According to IHU project [22] based on TCP, this study have modified and enhanced IHU to turn out a new framework with PR-SCTP called SCTP-IHU that performs audio/video streaming over IP in parallel. In SCTP-IHU framework, communication can seamlessly switch from the wireless network (IEEE 802.11b/g) to wired network (wired Ethernet) when wireless network is disconnected. When wired

network fails, this can be done seamlessly from wired network to wireless network as well. Timed PR-SCTP is employed to realize real-time transmission between two hosts. In addition, the multi-streaming let host send video, audio, and control packets over the streams independently.

3.1. **Software architecture.** The software architecture is shown in Fig. 5. We introduce the libraries which we use to development the system such as (a) an audio processing libraries SoundTouch [23], (b) a free codec for free speech Speex [24], and (c) a computer vision library OpenCV [25].

3.2. **System architecture.** As shown in Fig. 6, the dash-dot-line path is the transmission path of voice from microphone to network. We use ALSA to be our sound architecture and JACK as the audio server. The Recorder Module encodes the voice data by using Speex; the Call Module transmits the encapsulated voice data to network through Transmitter Module. The dash line path presents the receiving path of voice from network and the Receiver Module is responsible for receiving any data from network. In addition, it identifies the type of the received data. If the received data type is voice, the received data will be processed by Player Module after decoding by Call Module; afterwards, the voice is played by a speaker. As shown in Fig. 7, when the SCTP-IHU receives data, the SCTP-Handler Module will record the time of the receiving time corresponding to the peer address. In addition, the SCTPIP-Handler Module will check the receiving time periodically. If the SCTP-IHU does not receive any data within certain period of time, the SCTPIP-Handler Module will search one of the available IP addresses to be the new primary address.

3.3. **Object tree in SCTP-IHU.** As shown in Fig. 8, an object tree is sketched to depict the objects that are designed in this study and used to achieve the objective of tasks for performing multimedia streaming and video/voice over IP. The function blocks originated from IHU project marked by color light-blue have been modified to improve voice streaming over IP whereas the other function blocks with color grey maintain their usability as usual. Besides voice, we have also created a few new function blocks indicated by color red for generating video streaming over IP.

3.4. **Fully-integrated and friendly GUI environment.** In SCTP-IHU, as shown in Fig. 9, we designed a complete graphic user interface to integrate many needed functions to configure user communication environment, which made user use this software easily.

3.5. **Format of the defined packet.** As shown in Fig. 10, we design the new packet format for application layer, which is modified from the message format of IHU project [22]. Before we send the data to the remote node, we will fill the data into a packet. There are two segments in the packet format of our design, header and data. Header is used to describe the information of the packet, which is divided into SYNC pattern, packet size, INFO data and data length. SYNC pattern: The first three ASCII bytes of IHU packet compose the string: "IHU". Regarding DATA part, the size of this segment is variable (min 0, max 1016 bytes). This segment carries the data to be transmitted. It can be absent if the packet doesn't need to transmit any data. It can also be audio data or video data. The type of data transmitted is specified in the INFO header.

3.6. **Transmission using multi-streaming.** We transmit control/video/audio/special packet by taking different streaming number as listed in Table 2. When the application performs handoff procedure in order to avoid blocking with previous packet, the base streaming number will increase by 3. For instance, streaming number 2 is used to transmit video packet. After handover procedure, the video packet is transmitted by streaming number 5. In such a way, the lost packet number 3 will not block the following data, packet number 4 and packet number 5, as show in Fig. 11.

TABLE 2. The Streaming number for different packet type

Data Type	Streaming number
Control Packet	Base streaming number
Voice Packet	Base streaming number+1
Video Packet	Base streaming number+2
Handoff Packet	9

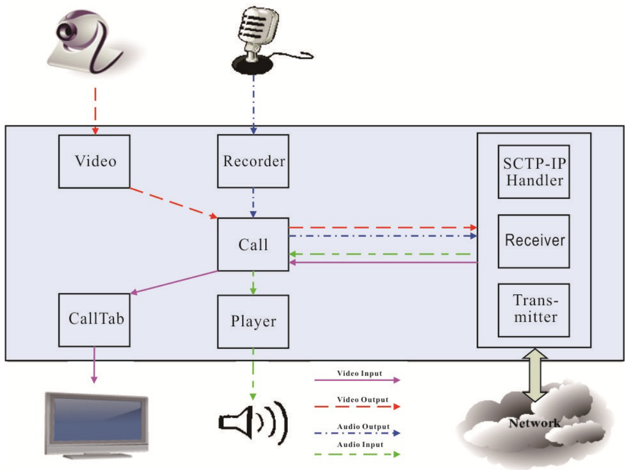


FIGURE 5. Software Architecture

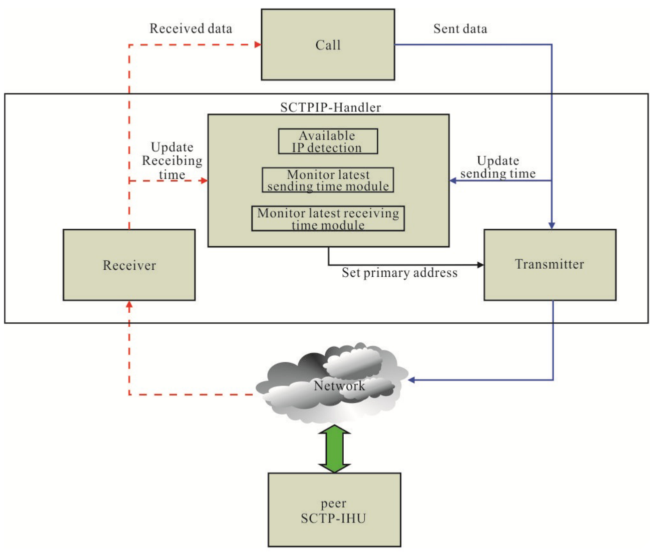


FIGURE 6. Architecture of SCTP-IHU

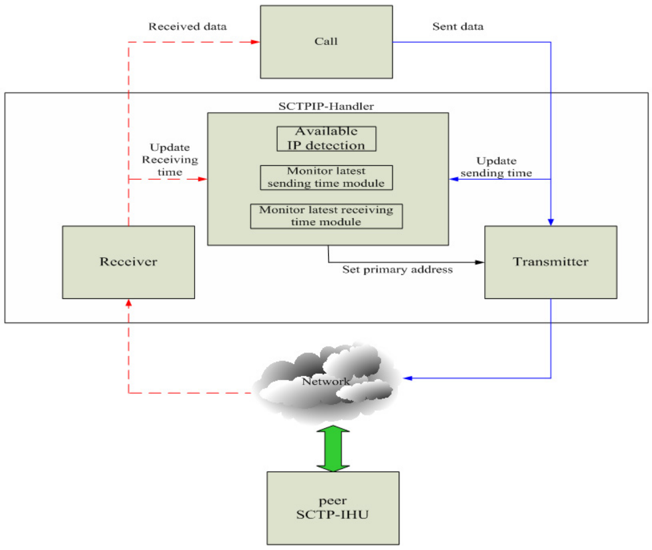


FIGURE 7. Block at right-hand side of architecture

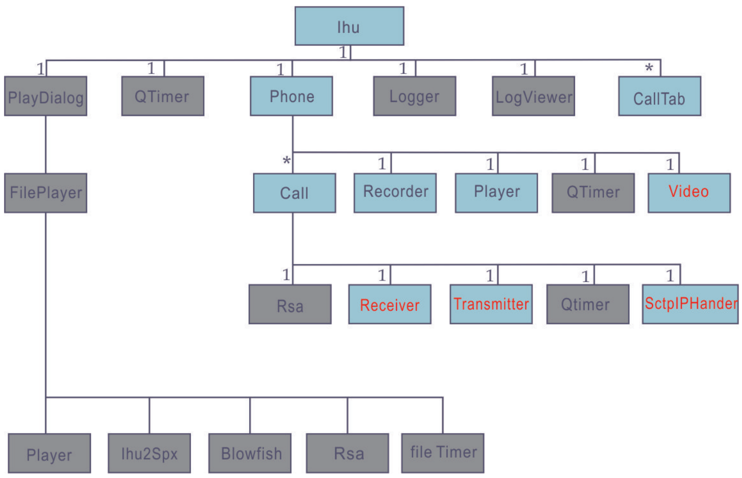
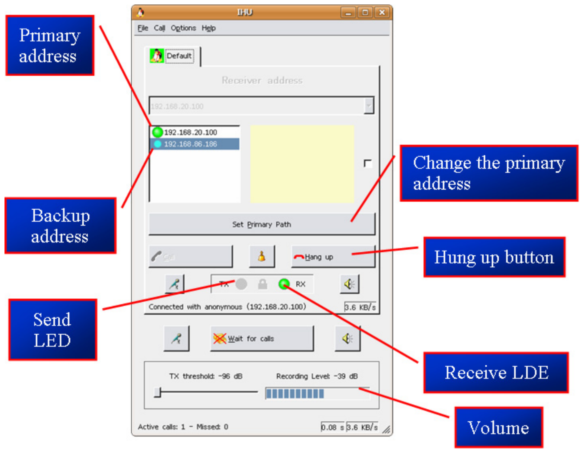


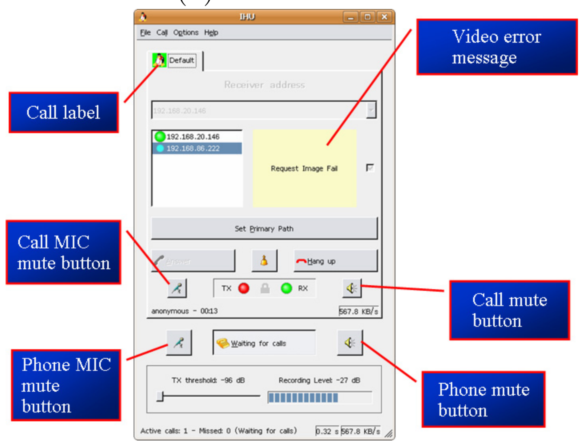
FIGURE 8. Object tree in SCTP-IHU

4. **Experimental results and discussion.** In the experiments, there are two Linux kernel based platform NAV270. Each of them equipped with four network interfaces, two 54 Mbps WiFi network interfaces and two 100 Mbps wired Ethernet network interfaces. We test the handoff between different paths by breaking wired Ethernet or WiFi network. When the application doesn't receive any data within threshold time, it will switch the primary address without re-connecting to the partner. In order to realize fast timed SCTP in voice/video over IP, we have implemented several handoffs between wired or wireless networks where system parameters are set listed below.

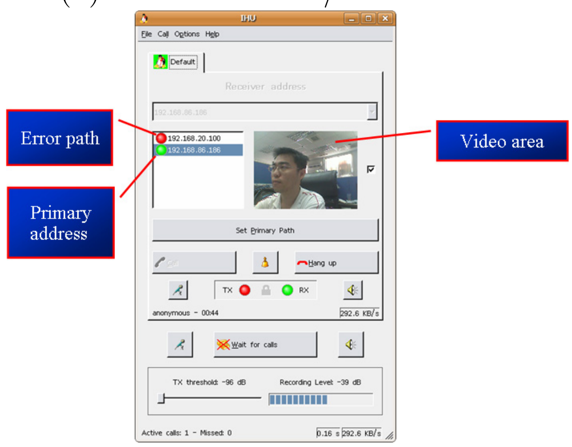
- Minimum retransmission timeout: 1000 mini-seconds
- Maximum retransmission timeout: 60000 mini-seconds
- Initial retransmission timeout: 3000 mini-seconds
- Maximum retransmission timeout during INIT: 3000 mini-seconds
- Maximum retransmission of INIT: 8 attempts
- Maximum retransmissions per address: 5 attempts
- Maximum retransmissions per association: 10 attempts



(a) Voice over IP



(b) Failure of video/voice over IP



(c) Success of video/voice over IP

FIGURE 9. User Interface of SCTP-IHU

Fig. 12 depicts the configuration of various handoffs in which we have categorized four types of handoff to switch the primary path (i) from wired to wireless network, (ii) from wireless to wired network, (iii) between wireless networks, and (iv) between wired networks. They are denoted as handoff_1, handoff_2, handoff_3, and handoff_4, respectively, in Fig. 12. For real time traffic underlying a timed PR-SCTP scheme, it is expected that some maximum tolerable delay will be set as a threshold to trigger a handover. This will involve setting the secondary path with the smallest delay to be the primary path. In order to investigate the threshold (the frame delay) for a fast handoff,

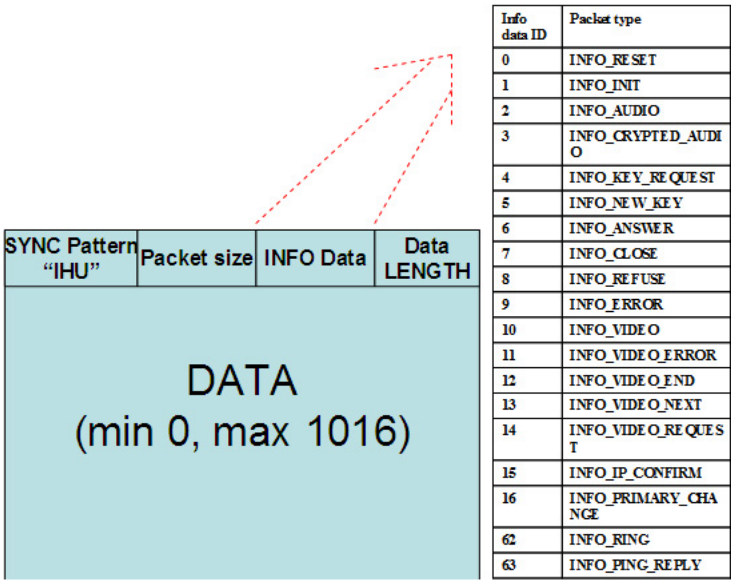


FIGURE 10. Format of a new defined packet

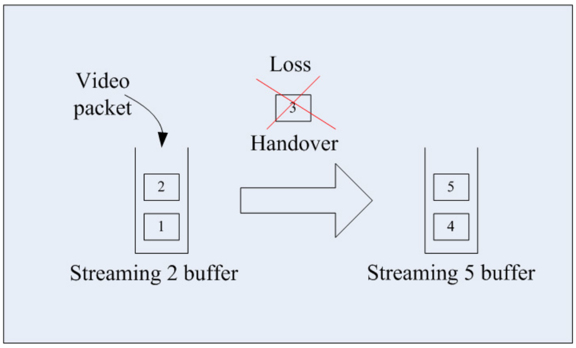


FIGURE 11. Changing streaming number for handoff

we have proceeded directly with the experiments and an example is described as clearly as possible below. In the event of the threshold with 4 sec, when the application does not receive within 4 sec, the application will perform the handoff procedure. We will detect the gap between the last frame of original path and the first frame of new path. As shown in Fig. 13, the gap is 4.622 sec as the threshold time is 4 sec, whereas SCTP default value for frame delay is 30 sec. Finally a summary of the results is listed in Table 3.

TABLE 3. The evaluation of gap for four types of handoff

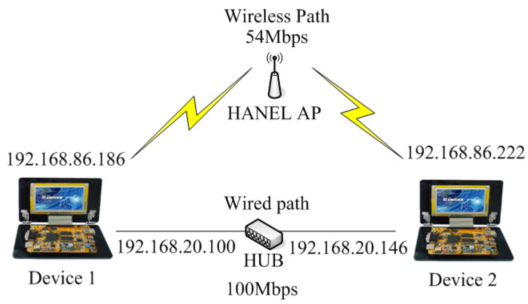
Threshold (second)	**gap (second)**					
	Handoff_1	**Handoff_2**	**Handoff_3**	**Handoff_4**	**Mean**	**Std.**
30*	47.849	47.135	48.274	46.689	47.487	0.710
4	4.584	4.518	4.538	4.509	4.537	0.033
3	3.346	3.341	3.374	3.322	3.346	0.021
2	2.534	2.521	2.556	2.515	2.532	0.018
1.5	1.865	1.843	1.874	1.829	1.853	0.020

*: SCTP default setting

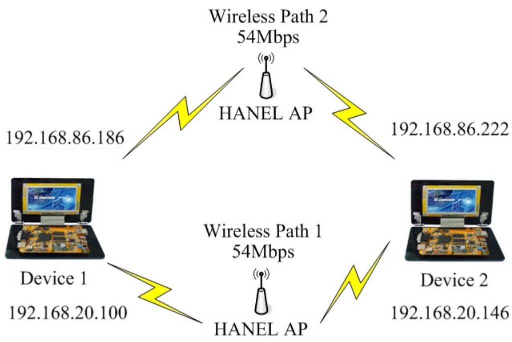
Unfortunately, the handoff procedure fails all the time when the threshold is less than 1.5 sec. This is because there is no enough time to initialize alternative path for continuing this association. As a result, if we do proceed to the handoff operation between heterogeneous networks, the suitable threshold time for a fast primary path switch is recommended 1.5 sec. Clearly in path switching when a handoff occurs the average gap is 1.853 sec with standard deviation 0.021 sec as shown in Table 3. Low latency and low packet loss are the two most critical design issues [1]. As listed in Table 4, this performance compared with the work USHA scheme [1] is definitely good enough to realize a seamless handoff between embedded platforms. Furthermore, we here stress that real-time streaming multimedia with PR-SCTP to achieve the least losses of transmitted packets, below 2

Table 4. The performance comparison

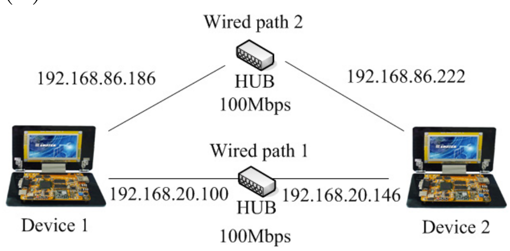
Performance	USHA	SCTP-IHU
Throughput	550.16 KB/s	1237.42 KB/s (Video), 117 KB/s (Audio)
Handoff Delay	2.5 sec	1.865 sec
Packet Loss Rate	3.26%	1.93%



(a) handoff from wired to wireless network and vice versa

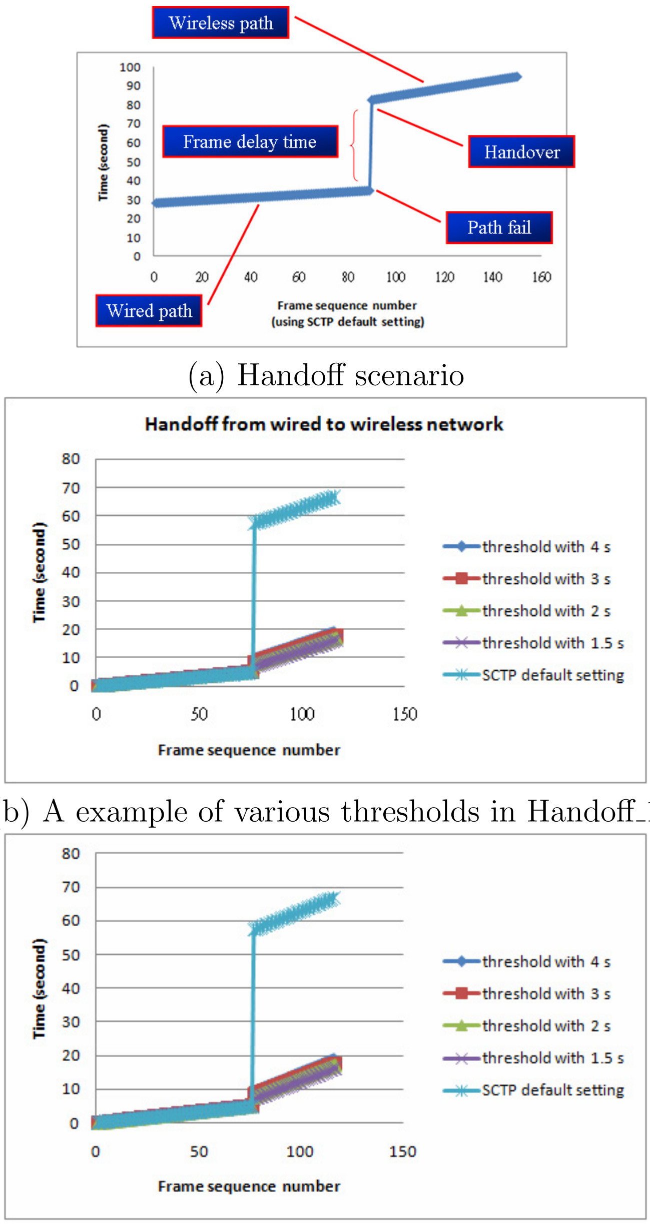


(b) handoff between wireless netwoks



(c) handoff between wired netwoks

Figure 12. Four types of handoff



(a) Handoff scenario

(b) A example of various thresholds in Handoff_1

(c) Comparison of handoffs

FIGURE 13. Handoff scenario for default and various thresholds in Handoff_1

5. **Conclusions.** This study has significantly improved the original IHU architecture to be a new framework called SCTP-IHU which provides a user-friendly control panel, redefines data packet format, enhances audio streaming manipulation, and creates a new feature of video streaming over IP. That is, our purposed SCTP-IHU provides a good application of fast streaming multimedia over IP. Instead of TCP or UDP, this study has exploited the features of PR-SCTP such as multi-streaming, multi-homing and partial reliability to overcome the crucial problems of the head-of-line blocking, handoff interruption, and non-real-time streaming multimedia, respectively. Therefore, seamless handoff between heterogeneous networks has been done successfully for streaming real-time voice/video over IP (V^2oIP). We have also explored frame delay for fast handoff and it turns out the minimum threshold is 1.5 s and the average gap is 1.853 s with standard deviation 0.021 s. With high throughput, low handoff delay, and minimum packet losses rate, our proposed approach outperforms the previous work TCP-based USHA scheme. The

future works will focus on the following sectors: file sharing, many-to-many voice/audio conference, and encode the video streaming.

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