The Design and Implementation of Messenger On-the-Drive

KAI-FONG CHOU, CHUNG-PING YOUNG, SHIOU-YU CHEN
AND LI-CHANG WANG

Department of Computer Science and Information Engineering
National Cheng Kung University
Tainan, 701 Taiwan

Telematics enhances the operations of a modern car. The Messenger on-the-drive (MOD) is developed as an inter-vehicle multi-networked voice/text communication utility, which enables the seamless wireless or telecommunications among the drivers and passengers in a fleet of cars. Two core features of Stream Control Transmission Protocol (SCTP), multi-streaming and multihoming, are applied for implementing the MOD project. We also use Partial Reliable SCTP (PR-SCTP), a characteristic extension of SCTP, to provide low latency handover in multi-network. By integrating our proprietary works and other open source projects, the MOD was successfully built.

**Keywords:** SCTP, partial reliable, VoIP, vehicle, multihoming

1. INTRODUCTION

In recent years, the functionalities of vehicles are getting more and more powerful. Various applications of vehicle-to-vehicle communication are introduced. In the Vehicle-to-vehicle communication environment, the network mobility is the most important issue, so we built up MOD [7] to replace the conventional radio communications with a multi-network system.

Based on IHU project [8], MOD is a vehicle-to-vehicle multi-networked voice/text communication system. It makes passengers be able to communicate with other passengers in another car through the wireless network smoothly without the delay. It can be applied for group trip, military operation, safeguard escort, and commercial transportation, etc., when a meet of vehicles moves together toward the same destination.

Comparing with conventional radio communications, the multi-network messenger provides three features. First, two-way communication is available on MOD, while the conventional radio is only one-way communication. The passengers in one car can easily have a conference with those who are in another car. The second feature is that the communication can seamlessly switch from the 802.11b to GPRS, when the 802.11b wireless network is disconnected. The communication among passengers will not be interrupted and they will not feel the lost of voice data packets. The third feature is that all passengers can also use PDA, Notebook, or other devices to access the Internet through the car pc in the multi-network communication.

We can imagine that Car PC will become standard equipment on every car in the future. There are a vehicular wireless network interface, which provides an unreliable and cheaper communication path, and a 3G network interface, which provides a reliable
and more expensive communication path, on every Car PC. In common sense, the passengers will communicate with each other by using vehicular wireless network interface in advance. When the distance between two cars is greater than a predefined threshold, the wireless communication will become unavailable. And MOD will change the path to the reliable communication path and maintain connection automatically and seamlessly. MOD will return the connection from a reliable path to an unreliable path, automatically, when connection of the unreliable link is available again.

The rest of this paper is organized as the following. In section 2, the SCTP and its extension is introduced. In section 3, we briefly describe the problem we encountered. In section 4, we describe the software architecture. In section 5, we list the innovative features of our project. In section 6, the design of the UI is depicted. In section 7, we briefly describe the limitation of MOD. In section 8, we list our ongoing work. In section 9, conclusion remarks are given.

2. BACKGROUND

In this section, we briefly describe the various fields involved in our work, highlighting key concepts.

2.1 SCTP

We use SCTP (Stream Control Transmission Protocol) [1, 2] in the vehicular environment. Due to the properties of multi-streaming and multi-homing, the SCTP can provide more reliable communication and better performance than TCP (Transmission Control Protocol) and UDP (User Datagram Protocol) in a dynamic environment. Multi-streaming allows data to be separated into different streams, so the serial data can be delivered to destination independently. Multi-homing [6] is a mechanism for an endpoint to support multiple IP addresses and multiple devices.

However, the SCTP handover mechanism [1, 2] is not suitable for our application. Because this mechanism is mainly used to judge whether the path is available or not by using heartbeat and heartbeat acknowledgement, and the changeover time of this mechanism is too long to be acceptable.

Therefore, in order to reach the requirement of real time interactive communication, we design another handover mechanism to make the changeover time as short as possible. We create an error threshold value to judge the packet loss, according to experiment results. If the quantity of packet loss is more than the error threshold, the handover, which we designed, will be invoked. And we confirm that the delay time is short enough to be ignored in our handover mechanism.

2.2 SCTP Extension

SCTP Partial Reliable Extension

In order to make the transmission real-time, we adopt the SCTP partial reliability extension. Partial reliability SCTP (PR-SCTP) allows a sender to skip unacknowledged messages when both endpoints support the extension. As of today the only partially reli-
able service specified is the timed reliability service.

Timed reliability means that the user can specify the lifetime of a message: when the lifetime is expired and the message hasn’t been acked yet, the sender stops the retransmission efforts and drops the packet. In the protocol control plane, the sender will send a forward TSN (Transmission Sequence Number), telling the receiver to move its cumulative ACK point forward. The effect of moving the ACK point forward is to consider the skipped messages as received and acked.

We place a time limit in our program on the life of any given message. When the time limit expires and the message has not been acknowledged, a skip message which specifies any stream and sequences that are skipped will be sent. We can make the voice communication partial reliable by this way. Moreover, partial SCTP can tolerate some packet loss and perform acceptable quality. This property is suitable to real time multimedia streaming such as our software.

Dynamic IP Address Reconfiguration

The feature of multi-homing allows a SCTP node has multiple IP addresses. There are some problems about IP configuration. In order to add and remove IP address dynamically, we apply one of the SCTP extension: dynamic IP address reconfiguration. It can provide a graceful method to add/delete to the interfaces of an existing association. We can dynamically add and remove the IP address even if the IP is used now by binding and connecting the IP address. The behavior can not affect the communication quality.

3. THE INITIATIVE HANDOVER MECHANISM

Under multi-network environment, low latency of handover is most important and necessary, and MOD is built with the spirit of handover seamlessly.

It makes communication switch seamlessly from the 802.11b to GPRS, when the wireless network is disconnected, and it will not let the passengers feel the delay or lost of voice data packets. Passengers can communicate with other passengers in another car through MOD smoothly without the delay.

In order to make sure handover seamlessly, we use SCTP to implement MOD. In this section, we describe the bottle neck of using SCTP’s failover mechanism to deal with the handover and the solution of resolving this problem by using PR-SCTP.

3.1 Problems of Using SCTP’s Failover Mechanism

SCTP has a built-in failure detection and recovery scheme, known as failover. Failover allows associations to dynamically send traffic to an alternate peer IP address when it is necessary. Thus, the one path is failed will cause chunk loss which triggers retransmission through the other path. SCTP uses a 4-SACK rule, in which the retransmission occurs on receipt of the 4th SACK that indicates the chunk is missed. The large amount of missing chunks caused by failed network connection will take long time to be retransmitted through the other link.

For voice streaming, in-time arrivals of data (chunks) are very important to have a smooth voice play without interruption. Moreover, SCTP also maintains a count across
all destination addresses on the number of retransmissions or Heartbeats sent to the remote endpoint without successful ACKs. The failure of the primary path may cause the counting number exceeding the configured maximum number, which results in the unreachable declaration indicating endpoint failure. Thus SCTP’s failover mechanism is not applicable in voice streaming handover.

At the first time, we consider we could use this failover mechanism of SCTP to change the primary path automatically and smoothly by tuning the SCTP parameter. But it does not work. Before receiver send ACK packet, sender’s buffer will queue the audio packets. When the path is broken, the sender’s buffer is not larger enough to accept all of the audio packets ceaselessly until receiver send ACK packet. And the slow start mechanism and retransmit mechanism will block all this association. At this time, the communication is blocked.

3.2 Handover Mechanism using PR-SCTP

After we are aware of these problems, we redesign the application layer protocol to transmit audio packet. We use partial reliable to replace original reliable protocol. And we write a SCTP event monitor to observe the audio packet error, while the error accumulates to threshold invoked our handover mechanism.

We also provide two policies to select reserve path in one association:

Roll polling in all paths – Our software will select reserve path by polling until the transmission successful.

User define path – In this policy, our software will select user define reliable path in the association. The scenario in this policy is the reliable path is more expensive than unreliable path. User could use unreliable path in normal times, but use reliable path while unreliable path failed.

4. HIGH LEVEL ARCHITECTURE

4.1 Software Architecture

Our data transferring could merge in one SCTP stack. The steps of voice packets processing are listed as below:

- SoundTouch audio processing libraries
  - Automatic changing the sound tempo, pitch and playback rate parameters independently [10].
- Speex (a free codec for free speech)
  - The Speex [9] codec uses the Ogg bitstream format, and the Speex designers see their project as complementary to the Vorbis general-purpose audio compression project.
  - Our software also supports the transmission of text messages and file data packets. The three kinds of data packets share the STCP stack by using multistreaming.
4.2 In the View Point of Network Communication

As shown in Fig. 2, in vehicle environment, there is not only one network communication interface on car PC. There will be multi-network when we embed the modules of network interface. There are 802.11b for short distance communication, and telecommunications network for far distance.

4.3 Speech

In the speech part, we use ALSA (Advanced Linux Sound Architecture), which is the default Linux sound architecture in the 2.6 kernel, as our sound architecture. ALSA’s main features include efficient support for all types of audio interfaces, fully modularized drivers, SMP and thread safety, backward compatibility with OSS and a user-space library to make application.

Furthermore, we employ JACK as the audio server. JACK is a low latency audio server and able to connect a number different applications to an audio device. For the audio compression, we use Speex which is unencumbered with the system. Speex can provide good quality speech and low bit rate and be robust to lost packets. Speex has a perceptual enhancement make the sound better when decoding.
4.4 Encryption

In the security part, we encrypt/decrypt the stream using a fast hybrid cryptographic system (RSA and Blowfish). RSA is a known public key encryption using two large prime numbers to compute the public key and private key. Blowfish is a variable-length keyed and symmetric block cipher. There is no effective cryptanalysis of Blowfish found until now. We use both encryption algorithms to make the system more secure.

5. UNIQUE AND IMPORTANT FEATURE

Fully Utilize SCTP Socket Power

From the publishing of RFC 2960 October 2000 to the first version of lksctp [11] published by Piggy in 2001, SCTP continues to be noticed by researchers. However, apart from simulating software, it has not been put into practice. (The result of searching sf.net are only nine pages in June 6, 2004.)

In order to let everyone use/observe the proceedings and capability of SCTP we hope that our software can demonstrate the above purposes.

Multi-network

We consider the multi-network will be popular in the near future! Every computer, cellular phone, or PDA will be equipped with more than one IP address and network interface (ex: 3G, GPRS, UTMS, IEEE 802.11bg, IEEE 802.11p, etc.). MOD will provide a better way for applications to fit with multi-network environment.

Good Policy with Handover

We provide two policies in our software. When MOD finds the present path can not be available any more, it will change the path to another one automatically and seamlessly.

Surely, user also can change the communication path manually. (According to our experience, the latency of resetting the path automatically is less than 1000 msec after MOD detect the error of the present path.)

Friendly User Interface

We provide a convenient graphic user interface. Users can dial, set, and add a new IP easily. We also provide complete security mechanism and a logger to record the behavior of user and application.

6. FULLY-INTEGRATED AND FRIENDLY GUI ENVIRONMENT

In MOD, we designed a complete graphic user interface to integrate many needed functions to configure user communication environment, which make user use this software easily.
6.1 Caller and Callee Function

As shown in Fig. 3, there are a “Receiver IP List”, a green “Call/Answer” button, a red “Hang up” button, a yellow “Ring” button (a button with a bell), a “Wait for Calls” button (a button with a telephone), a red “TX LED” and a green “RX LED” in Caller-call-callee Function. It provides a space for typing the IP address that you want to call. After writing the IP address of the computer that you want to call, click “Add” button to add the IP address to keep it in “Receiver IP List”.

![Fig. 3. Make a phone call.](image)

After adding the receiver address, click the green “Call/Answer” button to start the communication. MOD will ring the receiver and then, if he answers, it will capture and send your voice to the receiver and, then, you can receive the receiver’s voice. On the other hand, if you are waiting for calls (click “Wait for Calls” button). When you get a new call, this button will become “Answer” button. And you can click this button to answer the call. When you want finish the talk, you can click the red “Hang up” button to close the current communication.

6.2 Voice Controller

As shown in Fig. 4, there are two sliders and two buttons in the Voice Controller. The left slider is “Voice threshold”. When you don’t speak loudly enough to over the threshold, MOD won’t send any data that is considered silence. This function allows you to save network bandwidth when you are not speaking. You should change this value to a
suitable value so that TX led is off when you are not speaking. Of course this does not affect the data that you are receiving. And the right slider is “AGC (Automatic Gain Control) volume”, which controls the playback volume.

The left button with microphone is “Audio input control”. You can click this button to disable the audio capture or to mute your microphone. If you want to recover the audio capture, just click this button again.

6.3 SCTP Settings

As shown in Fig. 5, there are three blocks in the SCTP Setting. The first one is “Scpt Socket Options”. In this block, you can set the range of the retransmission timeout by set “RTO MAX” and “RTOMIN”.

![Fig. 5. SCTP setting dialog.](image)

The second is “Audio Packets”. Here you can choose to send voice/text packets with partial reliable or unreliable by enabling “Use partial reliable” or not. After enabling the “Use partial reliable”, MOD will send voice/text packets still continuously and will not be interrupted by ACK missing. Each packet has its own ACK signal, and there is a TTL (Time to Live) for each packet. You can set a suitable TTL for your voice/text packets. If MOD can not get ACK signal in TTL ms, it will print error message on SCTP monitor (as shown in 5.4).

The last one is “Error Handler (Automatic Hand Over)”. Here you can choose to allow changing to primary path automatically by enabling “Allow changing to primary path automatically”. After enabling “Allow changing to primary path automatically”, you can set a suitable “Error Threshold”. If error events are accumulated to “Error Threshold” times, MOD will invoke our handover mechanism.

Here we provide two handover policies for choosing, one is “Use default reliable path” and the other on is “Use polling”. If you choose “Use default reliable path”, when
error events are accumulated to “Error Threshold” times, the MOD will change path to which you select in the combobox. If you choose “Use polling”, when error events are accumulated to “Error Threshold” times, the MOD will change path to the next one.

6.4 Transmission Path Setting and Monitor

As shown in Fig. 6, there is a monitor for checking IP address and a “Set Primary Path” button under the monitor. After connecting with your partner, in multi-network case, you can check which IP address is used to transmit data or receive data through this monitor. And the “Set Primary Path” button is used to set primary path. You can choose an IP address which you want to set to be primary path, and then click “Set Primary Path” button to change the original path to it.

![Fig. 6. Status monitor.](image)

7. DISCUSSIONS AND LIMITATIONS

In this section, the scenario of the our handover mechanism is introduced. The problem of using SCTP’s default mechanism, which is called failover, for dealing with path disconnection is also investigated.

7.1 Does Not Provide Buddy List

Unlike most popular Instant Message software. MOD doesn’t provide “Buddy List” like functionality in this time. We consider use Jabbar protocol, a SIP base protocol, to implement use IP translation. And we’ll save user’s Buddy list that make user conveniently.

7.2 Other Limitation in MOD

No keyboard less support – This weak point make our software improper on vehicle.
No video conference support – But we will include video conference and broadcast in next version.

8. ONGOING WORK

8.1 Handover With GPS

We plan on creating a judgment mechanism to check when the unreliable path will
be nullified. And we decide to use GPS to get the locations of the cars and measure the
distance between two cars.

To begin with make cars carry with GPS to check the location of themselves. When
passengers use MOD to make connection and start to talk with each other, MOD can
exchange the location, which is received from GPS, from each other.

Then, MOD will measure the distance between two car and judge if the distance is
more than communication threshold. If it is, MOD will change the path to reliable path.

8.2 File Sharing

We plan on making MOD provides file transmission function to let passengers ex-
change files (ex: photo, music, etc.) with each other on the vehicle.

8.3 Video Conference

We plan on making MOD provide video conference to make passengers talk to each
other as face to face on the vehicles. This function will be more convenient and direct.

The requirements of the video conference implementation are less packet lost and a
wider bandwidth. Maybe we need to redesign an application layer protocol (ex: the
handover policy and the parameter of partial reliable).

8.4 Many-to-many Voice/Video Conference

MOD provides an one-to-one communication inter-vehicle only at present. And we
plan on using SCTP UDP STYLE to rewrite base socket function call, and redesign the
operation of graphic user interface to let user make many-to-many communications at the
same time.

8.5 Automatic Testing Unreliable Path

MOD provides the change automatically from unreliable path to reliable path. On
the contrary, it can’t change automatically from reliable path to unreliable path.

We plan to create a new mechanism to test the status of the unreliable path auto-
matically. When the unreliable path is connected and stable, MOD will provide two poli-
cies to reset primary path to unreliable path. One of the policies is hand-actuated by user,
and the other one is that MOD reset the primary path automatically.

9. CONCLUSIONS

In this paper, we present the internal design and implementation of the MOD 1.0.
Based on an enhanced handover methodology, the MOD 1.0 provides much better func-
tionality and performance than its predecessor and other product.

Due to its unique advantages, the MOD 1.0 was selected as silver winner at IOSESC
(International Open Source Embedded Software Competition), held in Taiwan, USA,
from 02/24/2005 to 08/21/2006.
The result also shows the average time consumed to complete voice streaming handover in our experimental network environments is about 1 second, which is much smaller than that when applying SCTP’s built-in failover mechanism.

ACKNOWLEDGEMENTS

We are very grateful to the OSSF (Open Source Software Foundry) for giving us a chance to attend the IOSESC (International Open Source Embedded Software Competition) and awarding the Silver Prize to us. We also appreciate the judges for their detailed and valuable comments, which make our product much better than original version.

REFERENCES


Kai-Fong Chou (周凱楫) was born in Taipei, Taiwan. He received the B.S. degree in Computer and Information Science at National Chiao Tung University in 2005. He is currently the M.S. student at the Computer Science and Information Engineering of Nation Cheng Kung University. He joined the NCTUns team and developed NCTUns’s GUI from 2001 to 2004. His researches mainly interests include embedded Linux, real-time OS, network simulation, inter-vehicle communications and robotics. His interests include coffee roasting, softball, and hacking open source project.
Chung-Ping Young (楊中平) was born in Taipei, Taiwan. He received the B.S. degree in Electronic Engineering from Chung Yuan Christian University, Taiwan in 1985 and the M.S. and Ph.D. degrees in Electrical Engineering from the University of Missouri at Columbia, U.S.A. in 1994 and 1997, respectively. After working for five years as a software engineer in PC and consumer electronics industry in U.S.A., he joined the Department of Computer Science and Information Engineering, National Cheng Kung University, Taiwan, in 2003, where he is currently an Assistant Professor. His fields of interest include real-time embedded systems, telematics, wireless sensor networks and virtual instrumentation.

Shiou-Yu Chen (陳秀瑜) received the B.S. degree in Computer Science Engineering at Tatung University in 2004, and received M.S. degree in the Computer Science and Information Engineering of National Cheng Kung University in 2007. Her researches mainly focus on embedded Linux, intervehicle communications and vehicle safety. Her hobbies are swimming, golf, and watching baseball game.

Li-Chang Wang (王立昌) received the B.S. degree in Computer Science and Information Engineering at National Cheng Kung University in 2005. He is currently the M.S. student at the Computer Science and Information Engineering of National Cheng Kung University. His researches mainly focus on vehicle safety, inter-vehicle communications, and embedded systems.