International Conference on Computer, Communications and Information Technology (CCIT 2014)

Railway Freight Dispatching Telephone System Based on VoIP in Wireless Networks

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Abstract - Railway freight dispatching telephone system is a kind of railway dispatching telephone systems, providing freight coordinating communication service. It requires high transmission data rate, high interactivity and customizable service in a high mobility environment. VoIP in wireless networks is a promising technology. It is significant to design a railway freight dispatching telephone system based on VoIP in wireless networks (IEEE 802.11 and IEEE 802.16). This paper analyzes the requirements of VoIP-based railway freight dispatching telephone system, proposes VoIP architecture for railway freight dispatching telephone system, analyzes the QoS performance of VoIP-based railway freight dispatching telephone system, and deployed experiments on a 20km railway freight line testbed.

Index Terms - Wireless Local Area Network (WLAN), Worldwide Interoperability For Microwave Access (WiMAX), Quality Of Service (QoS), Railway Freighting Dispatching Telephone System, VoIP.

1. Introduction

The railway dispatching telephone system is a very important component for railway dispatching coordination and management. The railway freight dispatching telephone system is a kind of railway dispatching telephone systems, providing freight coordinating communication service. It is smaller than passenger dispatching telephone system in scale, only including four kinds of users, dispatchers, station orderlies, locomotive drivers, and ground staff. It requires high transmission data rate, high interactivity and customizable service in a high mobility environment.

Voice over Internet Protocol (VoIP) is one of the most important technologies for next generation wireless communication systems. It can provide customizable voice service for the railway freight dispatching telephone system.

In recent years, wireless networks have been widely deployed. IEEE 802.11 wireless local area networks (WLAN) provide wireless Internet connection in many places not only in buildings but also in vehicles. IEEE 802.16 Worldwide Interoperability for Microwave Access (WiMAX) is known as Beyond Third Generation (B3G) communications technology for broadband wireless access. Unlike wireless LANs, WiMAX networks incorporate several Quality of Service (QoS) mechanisms at the Media Access Control (MAC) level for guaranteed services for data, voice and video [1]. WiMAX network can provide high transmission data rate and high interactivity for railway freight dispatching telephone system.

It is significant to design a railway freight dispatching telephone system based on VoIP in wireless networks. We analyzed the requirements of railway freight dispatching telephone system, proposed VoIP architecture for railway freight dispatching telephone system, implemented performance analysis in simulation and testbed.

The rest of this paper is organized as follows. Section II outlines requirement specification of railway freight dispatching telephone system. A VoIP architecture of railway freight dispatching telephone system is proposed in Section III. Section IV gives the QoS and performance analysis. The testbed is introduced in Section V and experiment results are provided in Section VI. Finally, Section VII concludes the paper.

2. Requirements of Railway Freight Dispatching Telephone System

Unlike traditional VoIP services, the VoIP services for railway freight dispatching telephone system must satisfy the following requirements:

1) *Multi-level priority*: Different call priority levels.

2) *Voice group call service*: The system can set and initiate a number of different priority group calls simultaneously. A client, who can be set in different priority group calls, is connected to the highest priority group call when more than one group call occurs at the same time.

3) *Force insert*: A higher priority call can add in an existing lower priority call. [2]

4) *Force disconnect*: The dispatcher which has the highest priority can set up a new call with each client in lower priority call without blocking.

5) *Dialogue recording*.

Furthermore some important performance requirements must be satisfied, e.g. real-time access to the information about online callings and locations of locomotives, high-speed wireless access to the network for mobile locomotives, large capacity of the dispatching telephone system, multi-line of individual calls and group calls, efficient call processing competence which is represented as Busy Hour Call Attempt (BHCA) times, overload control to ensure that important users (dispatchers) can communicate smoothly, and supporting QoS requirements of voice.

3. Proposed VoIP Architecture of Railway Freight Dispatching Telephone System

To fulfill above requirements, VoIP architecture of railway freight dispatching telephone system is proposed, including reference protocol stack and function structure.

A. *Reference Protocol Stack*

The reference protocol stack for the network of VoIP-based railway freight dispatching telephone system is shown in Figure 1.

Physical layer and data link layer are based on IEEE 802.16 (WiMAX) and IEEE 802.11 (Wi-Fi) for mobile wireless user-ends, internal wireless user-ends (in a locomotive or a building), and other network devices. Network layer and transport layer remain IP and TCP/UDP protocols. At application layers, there are five primary protocols. Real-time Transport Protocol (RTP) [3] is for transporting real-time data and providing QoS feedback. Real-Time Transport Control Protocol (RTCP) [3] provides out-of-band statistics and control information for an RTP flow. Session Initiation Protocol (SIP) [4] supports establishing and terminating multimedia communications, user location, user availability, user capabilities, session setup, and session management. Session Description Protocol (SDP) [5] is for describing multimedia sessions. Voice codecs are narrowband codecs G.7xx (for experimental analysis and compatibility mode, the system used some of them).

However, not all devices in the network must implement the whole protocol stack in Figure 1. For example, user-ends need the whole protocol stack since they must provide telephone service. While base stations only need lower three layers in the protocol stack since they provide access service.

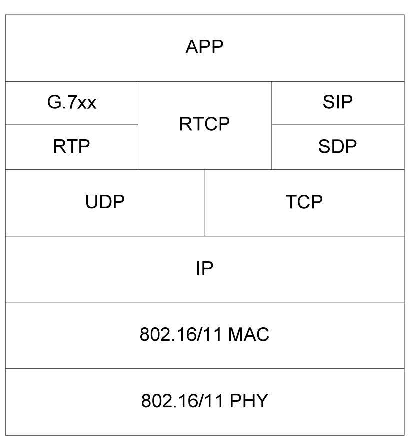


Fig.1 Reference protocol stack for the network of VoIP-based railway freight dispatching telephone system

B. *Function Structure*

The function structure of the VoIP-based railway freight dispatching telephone system is shown in Figure 2. It consists of Dispatch Telephone Service Center (DTSC), Client Platform (CP), Call Control Platform (CCP), QoS Control Platform (QCP), and Network Management Platform (NMP).

1) *Dispatch Telephone Service Center*

DTSC provides telephone service for clients. DTSC consists of six components in Figure 2. Register Server provides client registration and management. Proxy Server provides call routing, authentication, authorization, address resolution, loop detection, etc. Presence Server maintains presence information of clients and sends status notification to every online client which will refresh display in buddy list. Location Server receives position messages from every mobile locomotive client, calculates the dispatcher that the locomotive belongs to and the current front station and back station of the locomotive, and stores these results into database for reuse by other components. Redirect Server directs some clients to contact an alternate set of URIs (Uniform Resource Identifier). Database manages data for supporting the system operation.

2) *Client Platform*

Client Platform consists of six components in Figure 2. Individual Call provides one-to-one calling and emergency calling. Voice Group Call provides voice group call service and emergency group call service. Force Insert/Disconnect implements force insert and fore disconnect service according to the call priority. Media Processing provides voice codec and dialogue recording. User Agent implements SIP session. GUI (Graphical User Interface) provides a convenient graphic interface for users.

3) *Others*

CCP provides real-time session management. QCP fulfills QoS requirements. NMP manages the access service network.

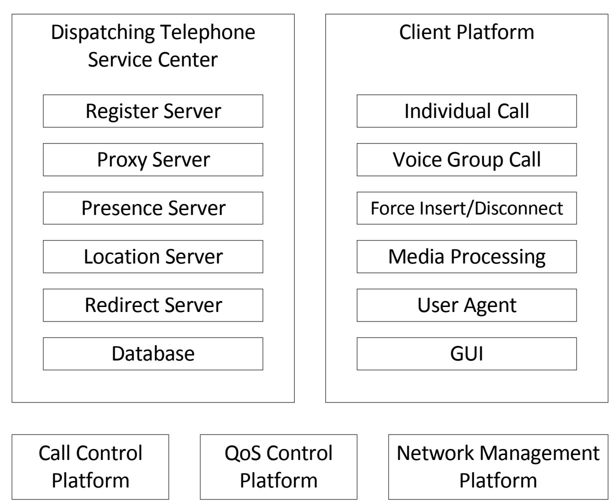


Fig.2 Function structure for the VoIP-based railway freight dispatching telephone system

4. QoS and Performance Analysis

A. *Capacity*

When designing a VoIP system, one of the most important issues is to determine the network voice capacity, in terms of the maximum number of simultaneous voice connections that can be supported in a wireless LAN (WLAN).

The IEEE 802.11 standard was originally designed to support best effort services in WLAN. It defines two modes of medium access control (MAC), distributed coordination function (DCF) and point coordination function (PCF). The standard also specifies two management modes, infrastructure and Ad Hoc modes. The WLAN in infrastructure with DCF is widely deployed. So evaluating the voice capacity of an IEEE 802.11 WLAN in infrastructure mode with DCF is of significant interest. The DCF used CSMA/CA (carrier sense multiple access with collision avoidance) to avoid collision. A station which needs to send a packet first senses the channel for at least a DIFS (DCF interframe spacing) shown in Figure 3. If the channel is idle, the station sets a random backoff counter value uniformly distributed in the range of [*0, CW*] (*CW* is contention window, which is initialized by CW_{min}). Then backoff counter is decremented by one for each slot when the channel is still idle. If the channel is busy before the counter reaches to zero, the decrementing process is frozen and is unfrozen until the channel is idle for a DIFS again. After transmission, the sender expects to receive an acknowledgement (ACK) within a SIFS (short inter frame spacing). If an ACK is not received with in $ACK_{timeout}$ period, the packet is considered lost. Then the *CW* is doubled until reaching CW_{max}. The station retransmits the packet *n* times until the ACK is received or *n* reaches retry limit value. When a packet is transmission successfully, the *CW* is reset to CW_{min}.

We have analyzed the theoretical capacity of VoIP traffic in IEEE 802.11 WLAN and derived capacity formulations for CBR and VBR VoIP traffic in [6].

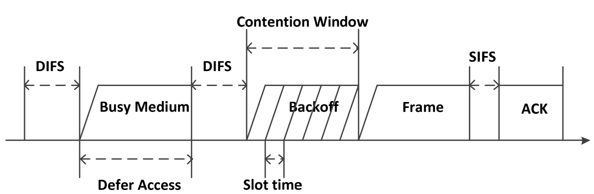


Fig.3 DFC MAC behavior

The CBR VoIP capacity without collision is defined as (1).

$$N_{CBR} = \frac{T_{PI}}{2\left(T_{DIFC} + T_{SIFC} + T_{header} + T_V + T_{ACK}\right) + T_{slot} \cdot CW_{\min} / 2} \quad (1)$$

In (1), N_{CRB} is the number of CBR calls, T_{PI} is the packetization interval. T_{DIFS} and $T_{S\ IFS}$ are the length of DIFS and SIFS. T_{header} is the time for transmitting the headers of RTP, UDP, IP, MAC, and PHY. T_v is the time for transmitting the raw voice packet. T_{ACK} is the time for sending ACK frame. T_{slot} is a slot time.

Using the parameters in Table 1 and Equation 1, the theoretical capacity of VoIP CBR traffic without collision is 16 calls.

In a WLAN unit, the capacity of 16 calls basically meets the needs of railway freight dispatching telephone system. In previous paper [7, 8, 9], they all discussed VoIP capacity improvements in IEEE 802.11e. The transmission opportunity (TXOP) in MAC layer can be a solution to improve the VoIP capacity.

Table 1 Parameters of IEEE 802.11b and VoIP

Parameters	**Size (byte)**	**Time (μs)**
Packetization interval		20000
DIFS		50
SIFS		10
PLCP preamble (shot)	9	72
PLCP header (shot)	6	24
MAC header + CRC	28	20.36
SNAP	8	5.8
IP header	20	14.55
UDP header	8	5.82
RTP header	12	8.73
G.711 voice data	160	116.36
ACK packet	14	10.18
Slot time		20
CW_{min}		31 (slots)

B. *Delay*

Delay is one of important QoS parameters in real-time system.

The Delay can be expressed like this:

$$T_{delay} = T_{codec} + T_{network} + T_{playback} \quad (2)$$

In (2), T_{delay} is the total delay. T_{codec} is the delay caused by the codec, which can be decomposed like this:

$$T_{codec} = T_{en} + T_{pack} + T_{de} \quad (3)$$

In (3), T_{en} is the encoding delay which is the time interval for encoding the voice data. T_{pack} is the packetization delay which is the interval for packetizing the encoded voice stream. T_{de} is the decoding delay which is the time interval for reconstruct the voice data. The codec delay is different depending on the selected codec mode.

In (2), $T_{network}$ is the delay caused by the network, which can be decomposed like this:

$$T_{network} = T_{tans} + T_{prop} + T_{mac} \quad (4)$$

In (4), T_{tans} and T_{prop} is the transmission time and the propagation time in network. T_{mac} is the delay caused in the MAC scheduling and queuing by network.

In (2), $T_{playback}$ is the playback delay induced by the playback buffer which is needed to smooth delay jitter.

5. Testbed and Experiment Results

To evaluate the performance of the VoIP-based railway freight dispatching telephone system over wireless networks, a demo VoIP-based railway freight dispatching telephone system is developed. A testbed is set up for some experiments.

A. *Testbed*

The architecture of testbed is shown in Figure 3. The testbed is deployed on a 20km railway freight line. There are one dispatcher, three stations, one locomotive, and several ground mobile cell phone. They are all installed appropriate VoIP client demo applets. In the IEEE 802.16 ASN, every station has a BS (base station) and a WiMAX RRM (Radio Resource Management) connecting to the Ethernet via an ASN-GW (ASN Gateway). Some of wireless clients access to IEEE 802.16 ASN via CPE (Customer-Premises Equipment). CPE can provide Wi-Fi for clients in stations and locomotives. In the locomotive, a device gets the position status messages from LKJ2000 (a locomotive running monitor device), and sends it to the dispatching telephone service center. The service center is installed the service center demo software.

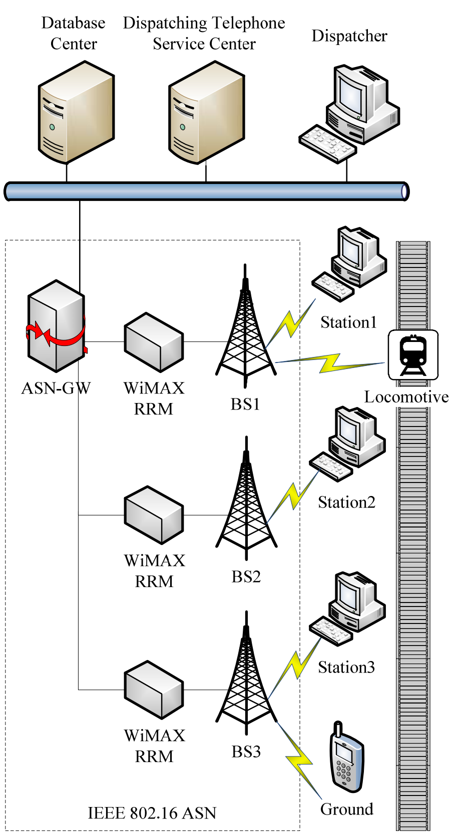


Fig.4 Architecture of testbed

B. *Experiment Results*

On the testbed, some experiments are implemented to analyze the performance of VoIP-based railway freight dispatching telephone system.

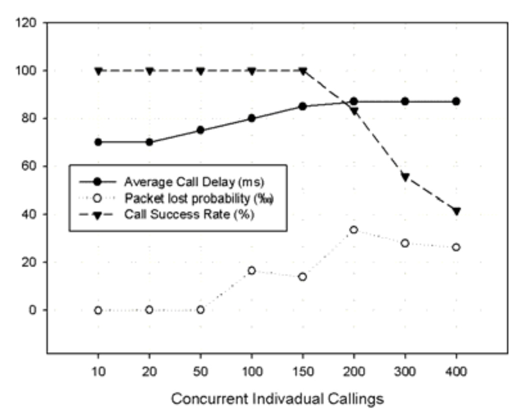
Table 2 shows the system performance on some key parameters. The results are from some experiments on the VoIP-based railway freight dispatching telephone system on the 20km test railway freight line. By comparing the actual situation of traditional railway freight dispatching telephone system working on this line, the VoIP-based railway freight dispatching telephone system can meet the performance needs.

Table 2 System Performance

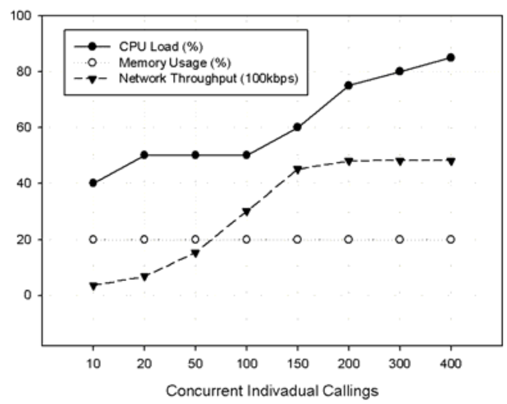
Parameters	**Experiment Result**
Number of Clients	1000
Max Number of Concurrent Individual Callings	150
Max Capacity of a Voice Group Call Service	150
Max Number of Concurrent Tripartite Callings[a]	40
BHCA Value	2.3×10^4/h
Average Call Delay	85ms
Call Delay Jitter	95%≤150ms

a. Tripartite calling is a voice group call service with three clients.

The performance of individual callings is shown in Figure 5. In Figure 5-a, Average call delay increases with concurrent individual callings between 10 and 150, and remains high value beyond 150. Similarly call success rate falls from 150. If the packet loss threshold is 2%, packet lost probability is below 20% when concurrent individual callings are not more than 150. So the maximal number of concurrent individual callings is 150. In Figure 5-b, When the number of concurrent individual callings reaches 150, CPU load and network throughput begin to rise. Memory usage stabilizes at 20%. So system resource can satisfy concurrent individual callings in VoIP-based railway freight dispatch telephone system.

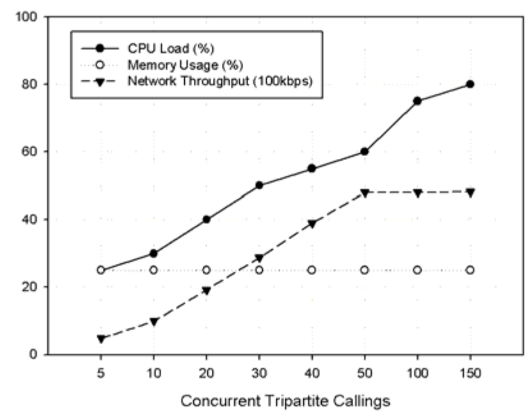


a. Average call delay, call success rate and packet lost in individual callings



b. System resource consumption

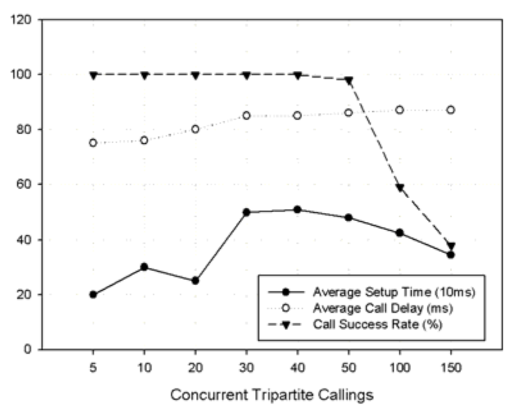
Fig.5 Performance of individual callings



b. System resource consumption

Fig.6 Performance of Tripartite callings

The performance of tripartite callings is shown in Figure 6. In Figure 6-a, Average call delay increases with concurrent tripartite callings between 5 and 40, and remains high value beyond 40. Similarly call success rate falls from 40. So the maximal number of concurrent tripartite callings is 40. In Figure 6-b, system resource consumption is like in individual callings. When the number of concurrent tripartite callings reaches 50, CPU load begins to rise fast; and network throughput is increased to a plateau value. Memory usage stabilizes at 20%. So system resource can satisfy concurrent tripartite callings in VoIP-based railway freight dispatch telephone system.



a. Average call delay, call success rate and packet lost in tripartite callings

From Figure 5 and 6, the performance bottlenecks in this telephone system are call success rate and packet lost probability. To improve system performance, we should find out some methods to increase call success rate and decrease packet lost probability.

6. Conclusions

This paper has analyzed the requirements of VoIP-based railway freight dispatching telephone system, proposed VoIP architecture for railway freight dispatching telephone system, analyzed the QoS performance of VoIP-based railway freight dispatching telephone system, and deployed experiments on a 20km railway freight line testbed.

Next, two aspects of research will continue: one is to improve the demo system for performance experiments, and another is researching QoS control to enhance system efficiency including handoff, capacity and call admission control.

Acknowledgment

The authors would like to thank the reviewers for their detailed reviews and constructive comments, which have helped improve the quality of this paper. This work was supported in part by National High Technology Research and Development (863) Program of China under Grant No.2012AA040912.

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