QPF SCHEDULING SCHEME FOR PERFORMANCE IMPROVEMENTS OF INTEGRATED MULTIMEDIA APPLICATIONS OVER 3.5G NETWORK

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Abstract: IMS can provide full IP technology platform which combines information and communications technologies to achieve the objectives of integrated multimedia service. In IMS, SIP is introduced to transmit session control signal. Currently, the session transmission time of SIP is affected by wireless channel bandwidth, Frame Error Rate (FER), message exchange volume, retransmission mechanism and other factors. In order to effectively reduce the delay time for session transmission, we did revise conventional MAC-hs scheduling algorithm (i.e. PF) and propose QoS-based Proportional Fairness algorithm, called QPF, to enhance the session setup on SIP with channel performance of wireless communication. We perform simulations with comparisons on two KPIs of Delay and Throughput between PF and QPF schemes, by using NS-2 software tool under various multimedia applications, such as VoIP, VoD and Web. The simulation results indicate that QPF lowers 20% for VoIP delay and obtains 8.19% more VoIP Throughput than PF. The QPF performance for VoD delay can be enhanced by 11.53%, the Throughput is increased by 8.05%. Also, the delay for Web of QPF can be improved by 28.58%, and the Throughput is enhanced by 25.25%. Consequently, the proposed QPF can enhance transmission performance and obtain a higher utilization of system resources for various multimedia applications in wireless network.

1 INTRODUCTION

IP Multimedia Subsystem (IMS) is the system used to control the multimedia application to be proposed by Release 5 (R5) of 3GPP. In IMS, the session transmission time for SIP (Session Initiation Protocol) refers to the session setup time plus VoIP data transfer time starting from User Agent Client (UAC) call to the receipt of ring back tone, in which the delay time can be divided into two parts: (1) network transmission time which refers to time spent for data transmission from one node to another, including transmission and retransmission time and (2) the delay time on network handling which is affected by Radio Link Control (RLC) and error detection. RLC signals will be affected by Multipath Fading, noise interference and buildings while transmitting over the wireless network. Also, this

will lead to worse wireless channel conditions and high error rate in wireless link layer. In order to reduce the impact of high error rate of wireless channels, this paper provides a more reliable data transmission service for the upper layer, with some guaranteed data transmission quality mechanisms such as Radio Link Protocol (RLP) introduced in the wireless link layer (Fathi et al., 2004).

In order to effectively control the wireless link layer over 3.5G network, a new sub-layer for MAC-hs (high speed) is added on the Media Access Control (MAC) layer of Node-B, in which the features of Fast Scheduling and Hybrid Automatic Repeat Request (HARQ) will effectively help reduce the delay time for network transmission, with different services quality that are defined by diverse media application demands as per delay time. The main purpose of this paper is to study and explore

the scheduling enhancement on 3.5G wireless network transmission to achieve Quality of Service (QoS) and performance improvement of multimedia application service.

The remainder of this paper is organized as follows. In Section 2, related literature review and discussion proceeded on the key technologies and applications of HSDPA and on new MAC-hs protocol. In Section 3, we will consider QoS in terms of PF in HSDPA and propose QPF (QoS-based Proportional Fairness) with scheduling to be designed to improve its retransmission and scheduling mechanism. In Section 4, we plan and set up simulation environment and list the execution procedure. In Section 5, we present and analyze the simulation results. In Section 6, we make the conclusion and indicate the future research direction.

2 PRELIMINARY AND RELATED WORKS

2.1 Integration of IMS and 3G Network

With the prevalence of communication system and Internet as well as evolution of new technologies, the integration of IP network and multimedia service has become a new trend for future development. IMS provides a good solution about how to attain real-time communications and multimedia applications over 3G networks (Kara & Planat, 2007). Hence, IMS is a technology platform upon SIP and is irrelevant to access network. Under a 3GPP IMS structure, Call Session Control Function (CSCF) is a primary component which is responsible for the control of SIP-based voice and multimedia conversation. Therefore, CSCF feature is described as followings and its structure is shown in Figure 1 (3GPP, 2010).

a) Proxy CSCF (P-CSCF) is the entry point of IMS. Also, it is responsible for subscriber network searching, and providing security and authentication functions.

b) Serving CSCF(S-CSCF) takes charge of phone service and session control, which is the control core of entire IMS.

c) Interrogating CSCF (I-CSCF) is used for every calling to select corresponding S-CSCF.

In Figure 1, Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) of GPRS pass internal IP network of 3G to provide packet service for communication signal, in which

the message flow for communication with traditional telecom network enters PSTN/ISDN via Media Gateway (MG) under the control of IMS. S-CSCF controls MG operation through Media Gateway Control Function (MGCF), while Signal Gateway (SG) supports Signalling System 7 (SS7) transmit over IP network and provides signalling service between IMS and traditional telecom network.
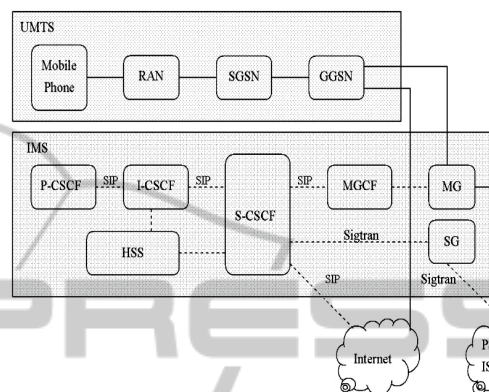

Figure 1: Basic infrastructure for the integration of IMS and 3G network.

2.2 Session Transmission Time of SIP

The session transmission time of SIP refers to SIP session setup time plus VoIP data transfer time. The session setup time was spent on the INVITE request sending from User Agent Client (UAC) to User Agent Service (UAS) which is informed of receipt of this session call from the Server (i.e. UAS receives the request of Acknowledgement (ACK). As a consequence, SIP time can be regarded as a total of completion of all message processing and delay time for VoIP data transmission between client-side and server-side. The delay for session setup involves many factors, in which the transmission delay over network is the major part including retransmission time delay and queue delay caused by FER. The procedure of SIP session transmission is shown in Figure 2, and the main steps are described as follows:

a) UAC issues an INVITE request carrying with media negotiation message at the beginning.

b) When UAS receives the invite request, it responds the session of 183 Session Progress.

c) When UAC receives the response, it issues PRACK temporary request as per UAS's media capability.

d) UAC proceeds to the verification with 200 OK.

e) UAS starts to issue 180 ringing message when resource reservation of UAS side is done.

f) UAS will issue 200 OK to respond to the primitive INVITE request.

g) UAC further confirms with ACK to be a final response to the INVITE request and then proceeds VoIP data transmission.
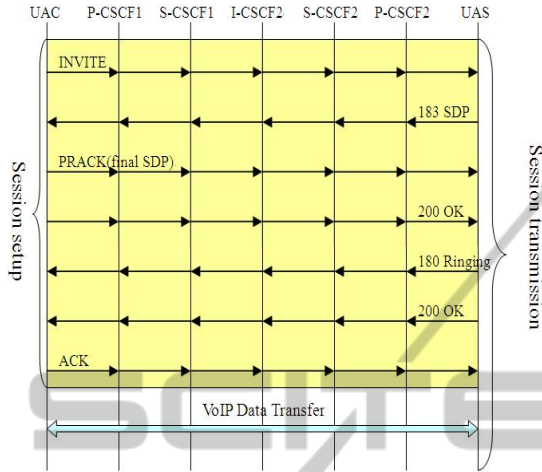

Figure 2: Delay time for SIP session setup.

2.3 Key Technologies of HSDPA

High Speed Downlink Packet Access (HSDPA), namely the 3.5G network, is an enhanced wireless technology of Wideband Code Division Multiple Access (WCDMA) (Krause, 2002). The objective of HSDPA technology is to slightly adjust or upgrade the software without changing current 3G/WCDMA network structure, to enhance Peak Data Rate of Downlink transmission packet data to over 10 Mbps and to shorten the packet latency for allowing the system to transmit packet data more efficient.

HSDPA technology lists diverse features including New MAC-hs and Channel, HARQ and Fast Scheduling to be compared to WCDMA (Ghosh et al., 2004). These features are illustrated as follows.
(a) New MAC-hs and Channel: HSDPA defines a new MAC sub-layer as MAC-hs (3GPP, 2001) on Node-B, where Node-B is a term used in UMTS equivalent to the BTS (base transceiver station), with a structure as shown in Figure 3. The MAC-hs takes charge of directing and distributing any a requested User Equipment (UE) for sharing channel resource of HSDPA and provides scheduling task of High Speed Downlink Shared Channel (HS-DSCH). The Transmission Time Interval (TTI) of HSDPA is 2 *m*s. It obtains five times growth comparing to TTI (10 *m*s) of Release'99 specification and which leads to an achievement of fast scheduling. The four functions of MAC-hs include: (1) Flow control; (2) Scheduling

with Priority handling; (3) HARQ; and (4) Transport Format and Resource Combination (TFRC) selection.
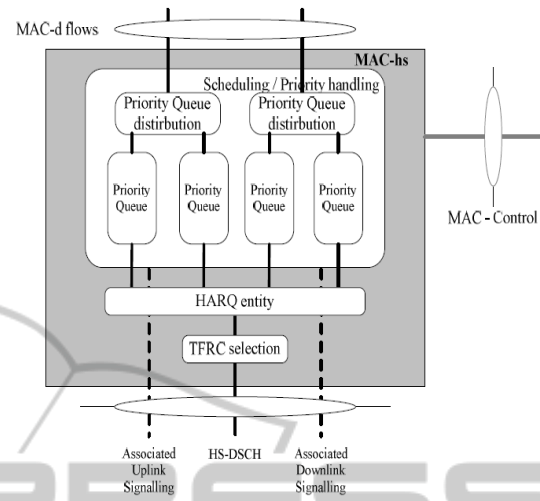

Figure 3: Structure diagram of MAC-hs at URTAN side.

(b) HARQ: The HARQ is an error correction technology that is frequently used in communication. It is a technology combining both Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ). Also, it can swiftly adjust the transmission rate of channel to achieve the combinations of FEC and retransmission according to the link conditions.
(c) Fast Scheduling: The packet scheduling of HSDPA is directly controlled by MAC-hs of Node-B instead of by RNC (Ameigeiras et al., 2004). The scheduling algorithms of HSDPA were normally divided into three types (Jalali et al., 2000): (1) Round Robin (RR), (2) Max C/I and (3) Proportional Fair (PF). When considering the shortest delay time for real-time voice transmission and network fairness of QoS, we choose to revise PF scheduling algorithm. User who gains an improvement on Channel Quality Indicator (CQI) will be able to benefit from higher priority rendered by PF. PF scheduling adopts an updating mechanism on average speed rate to avoid starvation phenomenon and to improve system delay time (Aniba & Aissa, 2004). In PF algorithm, each user will be given a corresponding equation with priority level P_i as shown below:

$$P_i = \frac{DRC_i(t)}{R_i(t)} = \frac{(C/I)_i(t)}{R_i(t)} \quad i=1,\dots,N \quad (1)$$

In Equation (1), DRC represents Data Rate Control while $DRC_i(t)$ refers to data rate supported by scheduled user i (Total N users) at time t according to current channel carrier to interference

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ratio. The updated equation of average transmission rate $R_i(t)$ introduced by user i as Equation (2).

$$R_i(t+1) = (1 - 1/tc) * R_i(t) \\ + 1/tc * Current\_Rate \qquad (2)$$

In Equation (2), where tc represents the length of Time slot while Current_Rate refers to the current frame transmission rate.

3 RETRANSMITTING AND SCHEDULING IMPROVEMENTS IN HSDPA

Retransmitting and scheduling of HARQ in HSDPA will be added on MAC-hs sub-layer, the major consideration lies in swiftly and clearly controlling usage conditions for radio resource, allowable for the packet to quickly transmit without any delay in order to enhance transmission efficiency. According to the conventional 3GPP, QoS parameters of Iub (Interface between RNC and Node-B) in HSDPA include (Holma & Toskala, 2006): (1) Scheduling Priority Indicator (SPI): the value range of SIP is [0, 1,…, 15], and the larger the number, the higher the priority level; (2) Guaranteed Bit Rate (GBR); and (3) Discard Time (DT). Also, we will categorize the media service types into Conversational, Streaming, Interactive and Background services by SIP priority upon the sensitivity level of services on the delay and diverse media service flow applications (Dixit et al., 2002; Li & Aaron Gulliver, 2009). The fundamental characteristics and application examples of various service types are summarized in Table 1.

Table 1: Class of media service in wireless network.

Service types	Conversational	Streaming	Interactive	Background
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream	Preserve time relation (variation) between information entities of the stream	Request response pattern Preserve payload content	Destination is not expecting the data within a certain time
Examples of the application	Voice、Video Phone (VoIP)	Streaming video (VOD)	Web browsing (Web)	Download of e-mails (FTP)

3.1 Design Issues of QPF Scheduling

The QPF is proposed to revise the scheduling and priority processor of MAC-hs based on HSDPA scheduling, using SPI indicator to determine those conversation types (e.g. VoIP) and higher FER frame which requires retransmission as higher scheduling weight than other service flows, followed

by streaming type, interactive type, and background type. The operational flow of QPF scheduling processing is shown in Figure 4 and its descriptions is illustrated in the following steps.
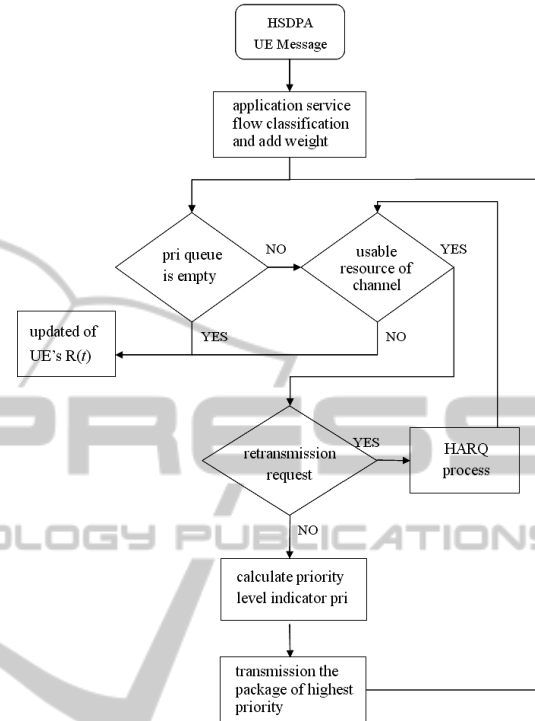

Figure 4: Operational flow of QPF scheduling.

Step 1: Calculation and Weight Setting of Priority Level
In Equation (3), W_i represents weight of service type of that subscriber, with rest of variables subject to Equation (1). In addition, the multimedia application type of 3G network is conformed to QoS needs, so that the weight of service type W_i is set as following: 4 for Conversational, 3 for Streaming, 2 for Interactive and 1 for Background.

$$P_i = \frac{DRC_i(t)}{R_i(t)} * W_i \quad i=1,\dots,N \qquad (3)$$

Step 2: Priority Value Queue
Place pri value of UE which stays synchronous with data transmission accompanied by Dedicated Physical Channel (DPCH) into the pri queue.

Step 3: Resource Allocation
Gather allocable real resources of Node-B.

a) Determine whether pri queue is empty and then judge whether there are usable HS-DSCH resources. The resource allocation process will come to an end if there are usable non-link channel resources or vacant pri queue.

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b) Calculate occupied real resources to be necessary and send Modulation and Coding Scheme (MCS) for UE according to retransmission request if HARQ requires a resending.

c) Dispatcher will firstly transmit a retransmission required for HARQ packet and calculate priority (pri) value as Equation (3).

d) Choose the user who gains highest pri value to execute dispatching, and then calculate average service rate of each separate users as Equation (2).

3.2 QPF Algorithm Design

Based on QPF operations as depicted in Figure 4, we design the QPF algorithm and list its pseudo codes as follows.

```
Algorithm QPF( )
{
Input: #Traffic flow of HSDPA UE
#include    "hsdpalink.h"    //MAC-hs
 implement-ation   (also    includes
 physical layer)
#include "cqi.h" //Lookup table for CQI
 to MAC-hs PDU size
#include    "error_model.h"    //Reads
 tracefiles  for  CQI  and  transmission
 power
#define  FLOW_CLASS():  //Traffic  flow
 classification and add weight
#define   FLOW_CHECK():   //Check   for
 activated flows
#define  SORT_POWER(): //Sort by Maximum
 relative CQI
#define  HARQ: //HSDPA Hybrid Automatic
 Repeat Request
#define CACULATE_DRC: //Calculate HSDPA
 LINK Data Rate Control
#define   struct   queue();   //define   a
 priority queue structure
static vector < float > power_; //HSDPA
 UE Power value
boolean retransmit_flag = true; //data
 retransmit indicator
char[15] flow_type
int weight
int flow_max
OUTPUT://  Loop  through  the  priorities
of each flow, in sort order
if  ( priority is Maximum)
    { Transmission the package }
else
    { waiting in queue buffer }
----------------------------------------
-----------------------
Method:
  BEGIN
  {
  Set weight = FLOW_CLASS();
  Execute FLOW_CHECK();
```

```
for (int i = 0; i < queue.size();i
 ++) {if ( power_.at(i) > 0.0 )
  { //channel resource enough
    if ( retransmit_flag )
       { EXEC HARQ ; break; }
      {power_at(i)=
            power_.at(i)* weight;}
  } else { power.push_back(0.0); }
  }
  EXEC SORT_POWER();
  EXEC CACULATE_DRC;
  }
  End
----------------------------------------
-----------------------
  Procedure FLOW_CLASS()
  {
  if (flow_type = "conversational")
   {set weight = 4}
    else if (flow_type = "streaming")
      {set weight = 3}
    else if (flow_type = "interactive")
      {set weight = 2}
      else if (flow_type = "background")
        {set weight = 1}
  Return weight;
  }End FLOW_CLASS()
----------------------------------------
-----------------------
  Procedure FLOW_CHECK()
  {
  for (int i = 0; i < flow_max; i ++) {
   if ( activated_.at(i) )
     { get power for this flow
     insert received power into
queue}}
  Return queue()
  }End FLOW_CHECK()
----------------------------------------
-----------------------
  Procedure SORT_POWER()
  {
  float pow_val;
  for ( int i = 1; i < queue_.size();
      i ++ ) {
  pow_val = power_.at(i);
  int j = i - 1;
   while ( j >= 0 && power_.at(j) <
     pow_val )
     {power_.at (j + 1) =
power_.at(j);
      j --; }
     power_.at(j + 1) = pow_val;}
   } End SORT_POWER()
----------------------------------------
-----------------------
}
END QPF.
```

4 ENVIRONMENT SETUP AND PROCEDURE OF SIMULATIONS

For the purpose of simulation experiments, this paper utilizes network simulator of NS-2.28 software version with insertion of EURANE-1.11 (Enhanced UMTS Radio Access Network Extensions) module to perform network simulations.

4.1 Setup of Simulation Environment

HSDPA network takes Dedicated Channel (DCH) and HS-DSCH into consideration. They are employed to signal control and data transmission, respectively. As a result, the status of packet transmission of these channels can be monitored to improve transmission efficiency of network. The network topology of simulation environment is shown in Figure 5.
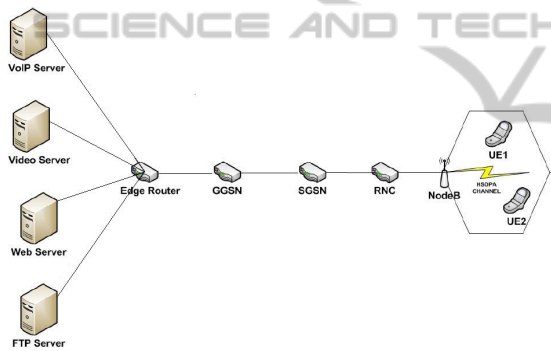

Figure 5: Topology of simulation environment.

Table 2: Settings of Simulated Parameters.

Service Types	Application	Traffic Source	Peak Sending Rate
Streaming	VoD	CBR	128 kbps
Conversational	VoIP	Expo. On/OFF	30 kbps
Interactive	Web	Pareto On/Off	60 kbps
Background	FTP	FTP	3434 kbps

In the experiment, this paper chooses a previous network application (Shreevastav et al.,2009) to stand for each media service flow: (1) *conversational* type: VoIP uses Exponential On/Off to generate data flow. (2) *streaming media* type: VoD uses CBR to simulate. (3) *interactive* type: Web uses Pareto On/Off to generate data flow. (4) *background* type: FTP application uses embedded FTP traffic generator of NS-2 to create information flow. The above related parameters setting is shown in Table 2. Because Nodes of HSDPA must be precisely configured before simulations, the configuration parameters of these nodes including bandwidth and delay are shown in Table 3.

Table 3: Parameters Setting of Nodes.

Starting node	Terminal node	Bandwidth	Delay
Serv. node B	Serv. node A	10 Mbps	35 ms
Serv. node A	GGSN	10 Mbps	15 ms
GGSN	SGSN	622 Mbps	10 ms
SGSN	RNC	622 Mbps	0.4 ms
RNC	Node-B	622 Mbps	15 ms

4.2 Procedure of Simulation

The core network topology is composed of GGSN, SGSN and RNC as shown in Figure 5. It will be allocated with different application service flows (2 VoIP, 2 VoD, 2 Web) by UE side. The FTP can be regarded as background flow. It will not only increase the number of FTP UE to measure the results of Delay and Throughput which pass the core network, but also compare their differences and analyze experimental results between PF and QPF schemes.

5 SIMULATIONS RESULTS AND ANALYSIS

In this section, we take Transmission delay and Throughput as KPIs (Key Performance Indicators) to measure the performance.

1. Transmission delay refers to time spent on network transmission from UE side to the server. To measure the results according to the transmission delay as defined by RFC 3550, two parameters will be derived when a voice packet (i) is received by the receiving side. They are packet timestamp Ri and packet RTP timestamp Si. Therefore, the transmission delay $D(i, j)$ between any two packet i and packet j can be determined by Equation (4).

$$D(i, j) = (Rj - Ri) - (Sj - Si) \qquad (4)$$

The experimental results as shown in Figure 6 indicate that QPF generally gains better efficiency than PF under light or heavy network traffic when there is a sufficient bandwidth. Practically, the QPF holds stable delays upon various application service flows. Therefore, simulation results demonstrate that the delay time can significantly decrease by taking the advantages of QPF algorithm.
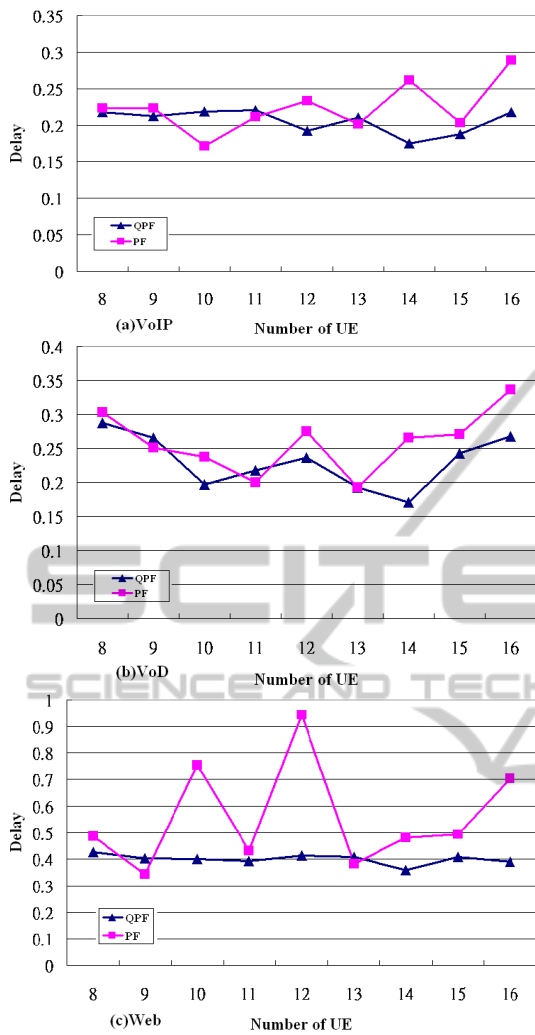
Figure 6: Simulation results in Delay.

Figure 7: Simulation results in Throughput.

2. Throughput is the average rate of successful packet delivery between two nodes of network. It is usually measured in bits per second (bit/s or bps), which is calculated as Equation (5).

$$Throughput = received\ bits\ /\ T \qquad (5)$$

In Equation (5), *received bits* represent the amount of bits t5o be received at all destination nodes during a certain period of time *T*.

According to the experimental results as depicted in Figure 7, the Throughputs of QPF are very close to those of PF under light network traffic (UE =< 11), but the QPF gains higher Throughput than PF under heavy network traffic (UE > 11). Therefore, simulation results indicate the Throughputs of QPF are better than PF in average.
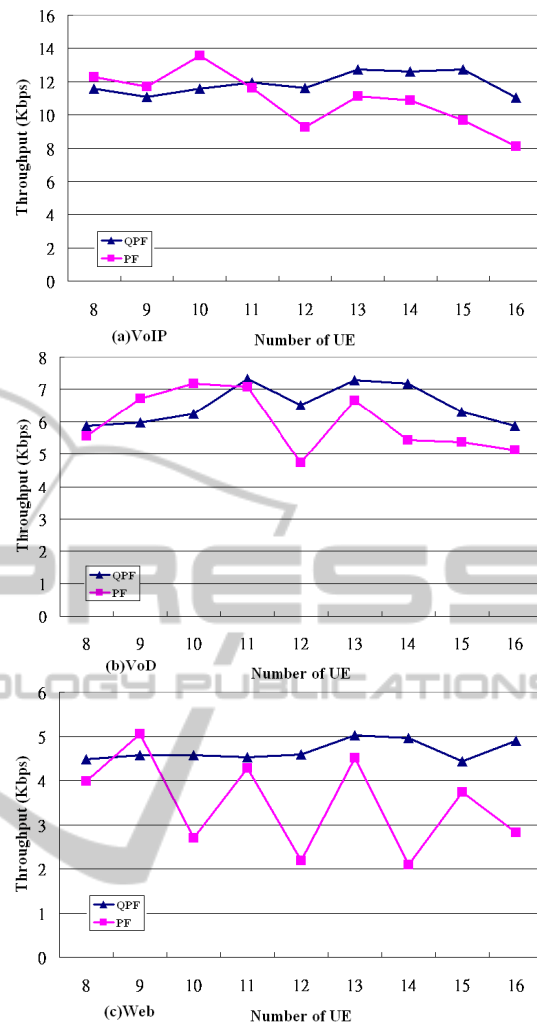
The comparisons of simulation results from Figures 6 and 7 are summarized in Table 4, which confirm our expectation that QPF can perform better than PF. According to Table 4, the sequence of delay time under QPF is Web (0.4) > VoD (0.23) > VoIP (0.2) from the weighted relationship for application service flows. It means that the VoIP service should be handled first because the VoIP gets the minimum delay. Also, the delay time of VoIP-qpf (0.2) is less than VoIP-pf (0.25) which is enhanced by 20%.

In addition, from the simulation results of Throughput, the QPF gains better efficiency than the PF in average. Under different application services, the QPF produces higher average Throughput than the PF in which VoIP service can be enhanced by 8.19%, VoD service by 8.05% and Web service by 25.25%. Hence, QPF is adoptable to multimedia application services, because of its advantages in

terms of Delay and Throughput performance over VoIP, VoD and Web application.

Table 4: Summarized simulation results.

KPI	Service	PF	QPF	% of diff.
Delay	VoIP	0.25S	0.2S	20
	VoD	0.26S	0.23S	11.53
	Web	0.56S	0.4S	28.58
Throughput	VoIP	10.9Kbps	11.9 Kbps	8.19
	VoD	5.89Kbps	6.51 Kbps	8.05
	Web	3.49Kbps	4.67 Kbps	25.25

6 CONCLUSIONS

IMS through full IP is a good solution to provide cross-platform integration service between different regions on heterogeneous wired or wireless network. However, in real-time transmission (e.g. VoIP), the quality of data transferring may be affected by a delay during network transmission. Service needs in diverse applications flows should be considered while transmitting signals over wired or wireless network.

This paper proposes the QPF with different levels of priority weights given as per diverse data type of application service flows such as VoIP, VoD and Web etc. We perform simulations with comparisons on two KPIs of Delay and Throughput between QPF and PF schemes, by using NS-2 software tool under various multimedia application services, such as VoIP, VoD and Web. The simulation results indicate that QPF lowers 20% for VoIP delay and gains 8.19% more VoIP Throughput than PF. The QPF performance for VoD delay can be enhanced by 11.53%, the Throughput is increased by 8.05%. Also, the delay for Web of QPF can be improved by 28.58% and the Throughput is enhanced by 25.25%. Consequently, the proposed QPF can provide better services to corresponding flow levels upon different applications over 3.5G network, which not only improves the effectiveness and efficiency of multimedia application services but also enhances the QoS of overall network.

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