LTE based telecommunication system for urban-guided transports

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Abstract

The traffic control and management of urban guided-transport system such as tramways and subways are based on several IT services having high data communication requirements, which are, today, carried over several different telecommunication infrastructures. In order to support existing as well as emerging IT services over a unique communication infrastructure, it is mandatory to ensure efficient QoS management that meets the requirements of management applications and especially the safety-critical ones. This paper presents an evaluation of the performances offered by a LTE (Long Term Evolution) based communication system for urban guided-transport. We focus on the performances of the QoS provisioning mechanism configured for the requirements of a set of signalling and management IT services. The performance evaluation is based on the well-known telecommunication simulator OPNET.

Keywords: Urban guided transport; LTE; CBTC; CCTV; QoS; OPNET.

Résumé

Le contrôle et la gestion du trafic des systèmes de transport guidés urbains tels que les tramways et les métros sont basés sur plusieurs services informatiques avec des exigences de communication de données élevées. Ces communications sont, aujourd’hui, supportées par des infrastructures de télécommunication différentes. Afin de supporter les besoins des applications existantes ainsi que ceux des applications futures, il est nécessaire d’assurer une gestion efficace de la qualité de service (QoS) et en particulier de garantir les exigences des applications critiques. Ce papier présente une évaluation du mécanisme de gestion de la QoS d’un système de communication pour les transports guidés urbains basé sur la technologie LTE (Long Term Evolution). Nous proposons une association entre un ensemble de services et des classes de QoS standards définies par la technologie. Nous présenterons une évaluation de performances réalisée avec le simulateur de télécommunication à événements discrets OPNET.

Mots-clé: Transports guidés urbains, LTE ; CBTC ; CCTV ; QoS, OPNET.

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Nomenclature

LTE  Long Term Evolution
CBTC  Communication Based Train control
CCTV  Circuit-Closed TeleVision
OPNET  Optimized Network Engineering Tool
QoS   Quality of Service

1. Introduction

To increase quality, reliability, safety and security, urban-guided transport systems rely on the deployment of several Information Technology (IT) applications and services that can be classified into three main categories: safety-critical applications (e.g., control-command), non-critical applications for train operation support (e.g., video surveillance, maintenance), and non-critical applications for infotainment (e.g., passenger information, internet on board). Since the middle of the 2000s, IEEE 802.11 networks with some proprietary developments are used for control and management of several metro systems such as in New York, Paris, Las Vegas, Lausanne, Singapore and Beijing (E. Kuun, 2004) (M. Fitzmaurice, 2005). However, the IEEE 802.11 has limited features and is not able to support communications of all applications using one management system. Therefore, it is usual today that a public transport operator deploys several wireless networks that are mostly non-interoperable, which leads to high operation and maintenance costs and a non-optimized radio spectrum use.

For a while now, the use of IEEE 802.11 technology for urban-guided transport systems is challenged with the growing success of the 3GPP LTE technology (IFSTTAR, 2012) which is the communication technology proposed by the 3GPP for the 4th Generation of mobile telecommunication networks. It targets a full-IP mobile telecommunication system with high data rate, low delay, improved coverage and spectrum efficiency. This technology is presented to be able to support heterogeneous traffics while ensuring QoS differentiation between them.

Taking into account all these promising performances, a communication system based on LTE will be able to match the requirements of urban-guided transport systems by accommodating all these applications in a single wireless access network while ensuring the primacy of safety-critical applications.

In this paper we are interested in the study of the QoS management mechanism proposed by LTE and its ability to ensure an efficient service differentiation between guided transport application categories. We investigate the performances regarding End-to-End (ETE) delay and packet loss, offered by LTE to these applications. The evaluation is based on the telecommunication simulator OPNET Modeler. We consider a realistic LTE deployment that carries a set of applications defined by urban-guided transport.

This work is performed within the SYSTUF project (IFSTTAR, 2012), which is a French research project that aims to demonstrate the feasibility of using a single communication system based on LTE to meet, simultaneously, the requirements of safety-critical and non-critical applications and to enable the development of innovative services contributing to seamless mobility which meets the growing demand for “smart and environment-friendly mobility”. It involves several industrials (Alcatel-Lucent, Mitsubishi-Electric, Alstom and Simpulse), research entities (Eurecom, IFSTTAR and Telecom Bretagne) and a public Transport operator (RATP).

The paper is organized as follows. Section II overviews the LTE technology. It describes the network architecture and the basic concepts of the QoS provisioning. Section III presents a set of management applications that may be defined by an urban-guided transport management system and their QoS requirements. Section IV presents a QoS configuration for an LTE deployment for a guided transport system. In section V, we present a simulation-based evaluation of communication performances offered to the applications. Conclusions are given in section VI.
2. LTE technology Overview

The LTE is the last progress of the Universal Mobile Telecommunications System (UMTS) standardized by the 3GPP. It proposes all-IP architecture (Fig. 1) that includes the evolution of radio and non-radio aspects of UMTS, represented respectively by the E-UTRAN (Evolved UMTS Radio Access Network) and the SAE (System Architecture Evolution), which covers the EPC (Evolved Packet Core) network.

The E-UTRAN ensures the radio resource management and the connectivity for User Equipment (UE). It offers radio coverage through a set of evolved NodeBs (eNBs), which are connected to the core network via the S1 interface. The EPC includes a set of management entities: the Mobility Management Entity (MME) responsible for mobility and session management, the Serving Gateway (SGW) responsible for routing and forwarding user data packets between E-UTRAN and the Packet Data Network (PDN) Gateway that provides gateway function to IP services such as IMS and/or Internet. In addition, the Policy and Charging Rule Function (PCRF) controls QoS policy and charging for users and services. The Home Subscriber Server (HSS) stores the users’ subscription data such as the QoS profiles.

LTE defines an ETE QoS management based on the concept of Evolved Packet System (EPS) bearer, which is used to identify packets belonging to a logical IP transmission path that receives a common QoS treatment between an UE and the PDN Gateway. In order to make the network able to support the service differentiation across services and users, LTE specifies a class-based QoS provisioning. There are two kinds of EPS bearers: Guaranteed Bit Rate (GBR) bearers and Non-GBR bearers. A GBR bearer has a specified data rates (UpLink (UL)/DownLink (DL) GBR values) for which network resources are allocated generally by an admission control function in eNodeB. Non-GBR bearer does not guarantee any particular bit rate.

Table 1. Standardized QCI (3GPP, 2008).

<table>
<thead>
<tr>
<th>QCI</th>
<th>Resources</th>
<th>Priority</th>
<th>Packet delay (ms)</th>
<th>Packet loss</th>
<th>Services</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>GBR</td>
<td>2</td>
<td>100</td>
<td>$10^{-2}$</td>
<td>Conversational voice</td>
</tr>
<tr>
<td>2</td>
<td>GBR</td>
<td>4</td>
<td>150</td>
<td>$10^{-3}$</td>
<td>Conversational voice (Live streaming)</td>
</tr>
<tr>
<td>3</td>
<td>GBR</td>
<td>3</td>
<td>50</td>
<td>$10^{-3}$</td>
<td>Real-time gaming</td>
</tr>
<tr>
<td>4</td>
<td>GBR</td>
<td>5</td>
<td>300</td>
<td>$10^{-4}$</td>
<td>Non-Conversational video (Buffered streaming)</td>
</tr>
<tr>
<td>5</td>
<td>Non-GBR</td>
<td>1</td>
<td>100</td>
<td>$10^{-6}$</td>
<td>IMS signalling</td>
</tr>
<tr>
<td>6</td>
<td>Non-GBR</td>
<td>6</td>
<td>300</td>
<td>$10^{-6}$</td>
<td>Video (buffered streaming)</td>
</tr>
<tr>
<td>7</td>
<td>Non-GBR</td>
<td>7</td>
<td>100</td>
<td>$10^{-3}$</td>
<td>Voice, Video (live streaming), interactive streaming</td>
</tr>
<tr>
<td>8</td>
<td>Non-GBR</td>
<td>8</td>
<td>300</td>
<td>$10^{-6}$</td>
<td>TCP-based (web, email, …), FTP, P2P, etc.</td>
</tr>
<tr>
<td>9</td>
<td>Non-GBR</td>
<td>9</td>
<td>300</td>
<td>$10^{-6}$</td>
<td></td>
</tr>
</tbody>
</table>
An EPS bearer is associated to a QoS Class Identifier (QCI), which is a pointer to a pre-configured set of node specific parameters (e.g. scheduling weights, admission thresholds, packet discard timer, etc.) that determines the packet forwarding behaviour to be provided to a service data flow delivered over this bearer. To ensure interoperability between operators and network equipment vendors, nine QCIs have been standardized by the 3GPP (c.f. Table 1).

Each QCI is characterized by priority level, packet delay budget and acceptable packet loss rate. The packet delay budget defines an upper bound for the time that a packet may be delayed between a UE and the PDN Gateway. The packet loss rate defines an upper bound for a rate of non-congestion related packet losses. In addition, a priority level is assigned for each QCI (the priority level 1 is the highest). The priority level and the packet delay budget determine how the Medium Access Control (MAC) scheduler, located in the eNodeB, handles packets sent over the bearers (e.g. a packet with higher priority is expected to be scheduled before a packet with lower priority).

Another parameter that characterizes an EPS bearer is the ARP (Allocation Retention Priority). It is used during call admission control to decide whether a GBR bearer establishment or modification request should be accepted or rejected in case of radio congestion. The ARP parameter contains information about the priority level, the pre-emption capability and the pre-emption vulnerability of a resource request. The priority level (from 1 to 15, where priority level 1 is the highest) defines the relative importance of a bearer request. The pre-emption capability and the pre-emption vulnerability are flags that define respectively, the ability of the bearer request to get resources that are assigned to another bearer with a lower priority level and the possibility that the bearer loses the resources assigned to it in order to admit another bearer with a higher priority level.

QoS in LTE is network-initiated where the operator offers a service to the subscribed UEs. QCI, ARP and UL/DL GBR are subscription parameters stored per-service in the HSS. These parameters are used by the PCRF to provide PCC (Policy and Charging Control) rules to the PDN Gateway. The PCC rules define packet filters that allow the PDN Gateway and the UE to distinguish packet flows and then match them to the appropriate EPS bearer. Once the PCC rules are executed, the associated EPS bearers are established. Service data flows, having the same QoS requirements (i.e. QoS parameters, QCI and ARP), are delivered over one EPS bearer. If this bearer is GBR, the PCRF adjusts the UL/DL GBR values to the sum of GBR values associated to each service data flow active on this bearer. Figure 2 shows the EPS bearer establishment procedure.

![Fig. 2. Establishment procedure of EPS bearers.](image)
3. QoS configuration for urban guided transport

The suitability of an LTE deployment for the urban-guided transport management will depend on the ability of its QoS provisioning mechanism to ensure the primacy of safety-critical services and to offer them guaranteed performances. A key issue for this is to propose an accurate mapping of the three categories of IT services to standardized QCIs.

3.1. IT services and their QoS requirements

For this work, we select a set of representative services and applications to evaluate the ability of an LTE deployment to ensure their needs. Table 2 summarizes the characteristics of data exchanges defined by these applications and their QoS requirements.

- **CBTC** is a control-command service for the automated driving of rolling stock. It relies on a bidirectional communication between on-board units and a wayside server, the Zone Controller (ZC). The latter provides the rolling stock with reliable information to adapt its operational status (e.g. speed) to the traffic situation. This application aims at controlling the distance between operating trains, in order to allow optimal use of the railway infrastructure while maintaining the safety requirements. CBTC is a safety-critical application, the cancelation of an on-going application or blocking a new application request is unacceptable.

- **Telephony** defines VoIP communications between rolling stock and the control centre, and between staff.

- **Discreet listening** defines the transmission of real-time audio (VoIP) from micros inside the rolling stocks to the management centre.

- **Voice announcements** consist in an audio streaming (VoIP) from management centre to an internal sound system on the rolling stock.

- **On-board surveillance** defines the transmission of real-time videos, from cameras inside the rolling stocks to a set of monitors in a control centre. It is based on the well known CCTV application. The traffic generated by this application is a constant streaming from one camera to a set of monitors.

- **Platform surveillance** defines the transmission of real-time videos, from cameras along the platform to a set of monitors in the driving cab. This service is also based on the CCTV application.

- **TV broadcasting** represents a real-time video streaming from the management centre to screens on the rolling stocks.

Table 2. A selection of IT services for urban-guided transport system.

<table>
<thead>
<tr>
<th>Service</th>
<th>Datagram size</th>
<th>Data Rate (kbps)</th>
<th>Packet loss threshold</th>
<th>ETE delay mean threshold</th>
<th>Jitter threshold</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBTC</td>
<td>200 bytes</td>
<td>UL 8 kbps</td>
<td>$10^{-3}$</td>
<td>50 ms</td>
<td>--</td>
<td>99.9%</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 4.6 kbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Telephony</td>
<td>160 bytes</td>
<td>UL 64 kbps</td>
<td>$10^{-2}$</td>
<td>150 ms</td>
<td>40 ms</td>
<td>--</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 64 kbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Discreet listening</td>
<td>160 bytes</td>
<td>UL 64 kbps</td>
<td>$10^{-2}$</td>
<td>150 ms</td>
<td>40 ms</td>
<td>--</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 64 kbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice announcements</td>
<td>160 bytes</td>
<td>UL 64 kbps</td>
<td>$10^{-2}$</td>
<td>150 ms</td>
<td>40 ms</td>
<td>--</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 64 kbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>On-board surveillance</td>
<td>1000 bytes</td>
<td>UL 2 Mbps</td>
<td>$10^{-3}$</td>
<td>100 ms</td>
<td>20 ms</td>
<td>--</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 2 Mbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Platform surveillance</td>
<td>1000 bytes</td>
<td>UL 2 Mbps</td>
<td>$10^{-3}$</td>
<td>100 ms</td>
<td>20 ms</td>
<td>--</td>
</tr>
<tr>
<td></td>
<td></td>
<td>DL 2 Mbps</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>TV broadcasting</td>
<td>1000 bytes</td>
<td>DL 6 Mbps</td>
<td>$10^{-3}$</td>
<td>100 ms</td>
<td>20 ms</td>
<td>--</td>
</tr>
</tbody>
</table>

3.2. Allocation of Standardized QCI

We allocate to standardized QCIs to the set of services based on their QoS requirements as shown in table 3.

The QCI 5 (recommended for IMS signalling) is allocated to the CBTC application. This Non-GBR QCI has the highest priority and a very low packet loss $10^{-6}$. With regards to delay, the value indicated by the LTE specifications is 100 ms. Remember that this value is an upper bound. Actual packet delays should typically be lower than the packet delay specified for a QCI as long as the UE has sufficient radio channel quality.
We identify the requirement of voice services as corresponding to QCI 1. This QCI is proposed by LTE specification for voice traffic and fits well their requirement (c.f. Table 1).

The CCTV services are associated to QCI 2, which involves a GBR bearer and meets CCTV requirements regarding delay and packet loss. The corresponding priority level is 4, which is lower than the QCIs proposed for the safety-critical service and the voice services.

The TV broadcasting is associated to the QCI 9. It has the lowest priority compared to other services. In addition, this is a non-GBR QCI and therefore it does not monopolize resources.

Table 3. QCI assignment to IT services.

<table>
<thead>
<tr>
<th>Service</th>
<th>QoS Class</th>
<th>Guaranteed Bit Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBTC</td>
<td>5</td>
<td>--</td>
</tr>
<tr>
<td>Telephony</td>
<td>1</td>
<td>128 kbps</td>
</tr>
<tr>
<td>Discreet listening</td>
<td>1</td>
<td>128 kbps</td>
</tr>
<tr>
<td>Voice announcements</td>
<td>1</td>
<td>128 kbps</td>
</tr>
<tr>
<td>On-board surveillance</td>
<td>2</td>
<td>8 Mbps</td>
</tr>
<tr>
<td>Platform surveillance</td>
<td>2</td>
<td>8 Mbps</td>
</tr>
<tr>
<td>TV broadcasting</td>
<td>9</td>
<td>--</td>
</tr>
</tbody>
</table>

4. Simulation and Results

We propose a simulation based performance study to validate the QCI mapping. We use the telecommunication simulator OPNET Modeler.

As an evaluation context, we consider a tramway line with equally spaced stations (every 1km). We consider a linear deployment of 15 eNodeBs with an inter-distance of 300m. The tramway is moving through a line 50m away from the eNodeBs’ line with a speed of 40km/h. At every station, the tramway stops for 30s. It operates continuously the following services:
- One CBTC
- One Telephony
- One Discreet listening
- One Voice announcements
- Four On-board surveillance
- and two TV broadcasting.

The service Platform surveillance operates when the tram is at station. Sixteen CCTV video flows are carried from cameras along the platform to the rolling stock.

The LTE deployment is configured at 5.9 GHz the frequency band with a 20 MHz bandwidth used in Time Division Duplexing (TDD) mode (UL/DL 3:2). Adaptive Modulation and Coding (AMC) is activated. The proposed configuration offers 50.1377 Mbps on the uplink and 30.0298 Mbps on the downlink. These values are estimated by the control admission function of OPNET’s LTE model.

Figure 3 shows the topology implemented in OPNET. The traway and stations are modeled as LTE UEs. The eNodeBs are connected to the EPC through an IP cloud. Behind the EPC, there are the application servers that model the management centre for IT services. We adopt the QoS configuration given in Table 3 (QCI and GBR values). CCTV and voice services have the same priority level regarding urban guided transport context. Therefore, they are assigned the same ARP parameter values (priority: 6, can pre-empt: yes, vulnerable: no). For such configuration, GBR bearers already defined are always admitted and cannot be pre-empted even if radio resources become insufficient.
The simulation runs for 625s. All applications start at 100s, except *Platform surveillance*, which operates only for 30s when the tram is stopped at stations.

We collect a set of statistics to evaluate the communication performances perceived by the IT services. We are interested in the ETE delays and received vs. sent traffic to all departments concerned.

### 4.1. Performances for the CBTC service

The ETE delay for the downlink traffic (c.f. Fig. 4.a) is around 1ms and reaches 10ms at stations. For the uplink, the ETE delays are around 15ms (c.f. Fig. 4.b). They are around 50 and 70ms at stations. The overall ETE delay average is 35ms. This average is limited to 15ms while considering values of non-overloaded network parts.

![Fig. 4. ETE delay of the CBTC service (a) Downlink; (b) Uplink.](image)

Figure 5 shows the sent vs. received traffic of the CBTC service at uplink and the downlink. The packet loss rate is equal to 0.96% in the uplink and 0.27% in the downlink. These values do not meet the requirements of the CBTC service.
Based on the observation of radio link statistics given by OPNET, we notice a radio conditions’ deterioration at the stations. As we rely on an adaptive modulation, the system steps down to a lower-order modulation scheme, thereby slowing throughput and increasing delays. This explains also the results for the uplink, which is not overloaded in stations.

4.2. Performances for the voice services

Voice services include the Telephony, Discreet listening and Voice announcements. The three services are similar regarding data traffic characteristics and QoS requirements. We focus here on results of the Telephony service.

Simulation results show that the ETE delay for voice services is, outside stations, lower than 120ms and the packet loss rate is around 0.38%. At stations, the ETE delay exceeds 140ms for some values and the packet loss rate is equal to 1.41% (c.f. Fig. 6). This does not correspond to the requirement of voice services, they require packet loss rate less than 1%. The explanation is the same as the CBTC service: a degradation of the radio channel that decreases the modulation order.

4.3. Performances for the surveillance services

Surveillance services including On-board surveillance and Platform surveillance services are similar regarding data traffic characteristics and QoS requirements. In the following, we focus on the results of the On-board surveillance service.
Figure 7 shows the ETE delay and jitter for all CCTV flows. The ETE delay average does not exceed 35ms outside the stations. There, it reaches several tens of milliseconds and exceeds 140ms. The maximum jitter is equal to 4ms and the packet loss rate is equal to 0.83%. As the previous services, the performances, outside the stations, fit the services requirements and the LTE QCI previsions. With an overloaded radio environment, the performances degrade and reach the defined thresholds. Results of Platform surveillance are worse as only 8 Mbps is guaranteed to carry 32 Mbps over one bearer.

4.4. Performances for the TV broadcasting

This service is considered as a best effort service in our configuration. Figure 8 shows the ETE delay (c.f. Fig. 8.a) and the received traffic statistics (c.f. Fig. 8.b) for the TV broadcasting service. Outside stations ETE delay is around several tens of milliseconds, less than the 100ms threshold required by the service (c.f. Table 2). Inside stations, with the increase of the load on LTE cells, the ETE perceived by this service increases significantly to several hundreds of milliseconds to reach 900ms. The packet loss rate average is 1.98%. These results show that the performances perceived by the TV broadcasting service are more damaged than the ones perceived by the CBTC, voice and surveillance services. This can be explained by the fact that they have been assigned to QCIs with higher priorities and guaranteed resources.

4.5. General observations

The results presented above show that the QoS provisioning in LTE works well with a reasonably loaded radio link. However, with an overloaded radio link, the thresholds specified by LTE and required by the studied services are exceeded. The required performances are exceeded punctually for the safety-critical service. For the
other services, the exceeding of the requirement is more important for services with less priority. The service associated to a QCI with non-guaranteed resources sees its performances remarkably exceeding the required threshold. This demonstrates that the LTE QoS provisioning mechanism is working, despite the shortcomings in overloaded radio links.

We believe that the radio link overload causes dysfunctions in the QoS policy implementation. These dysfunctions are due in part to possible radio interferences. In addition, it is possible that the scheduling algorithm that OPNET implements for the wireless link access becomes less effective in the case of degradation of radio conditions.

5. Conclusion

For a while now, the 3GPP LTE technology is presented as the communication technology of the future for the management of IT services in urban-guided transport systems. This is due to the evolved QoS provisioning mechanism that it defines. It is presented to be able to support heterogeneous traffics while ensuring QoS differentiation between them. In this paper, we proposed an evaluation of the performances that this technology can offer in an urban-guided transport use. We defined a basic scenario with a realistic wireless deployment and a set of urban-guided transport services. We proposed a mapping of these services to standardized QoS classes defined by LTE. We use the OPNET simulator for performance evaluations. Results have shown that the LTE QoS provisioning works well with a reasonably loaded radio link. However, we noticed several shortcomings in overloaded condition that cannot be acceptable in safety-critical use context. These shortcomings have to be studied deeply based on the more extensive evaluation scenarios.

In our current work, we are studying the behaviour of the QoS provisioning mechanism in overloaded environments. We are also working on the definition of new scenarios that involve a wider range of urban-guided transport IT services such as remote maintenance services, adaptive bit rate video services, etc.

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References


