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LTE Relay Architecture and Its Upper Layer Solutions

Xiaolong Huang, Fatih Ulupinar, Parag Agashe, Duncan Ho, Gang Bao

***Abstract*—the relay technology is an essential piece of LTE-Advanced in its pursuit of cost effective capacity enhancement and coverage extension. Simple RF repeaters have been widely used in cellular technologies to offer coverage extension. However, they cannot distinguish desired signals from interference and noise and are unable to operate with sophisticated mechanisms, such as power control, beam-forming, and adaptive modulating/coding. As a result, simple RF repeaters do not provide capacity enhancement nor do they reach the capacity potential of extended coverage areas. The use of more sophisticated relay in cellular technologies is relatively new. With its great benefits in QoS differentiation, capacity enhancement, and coverage extension, the development of LTE relay technology faces challenges in providing backward compatibility, minimizing complexity, providing sufficient QoS differentiation, reducing over the air overhead, and ensuring security. This paper addresses these challenges with solutions in the aspect of relay architecture and upper layer designs.**

I. Introduction

The approach of deploying numerous LTE [1] base stations (eNBs) with wired backhaul links to provide capacity enhancement and coverage extension becomes cost ineffective when the deployment targets to provide in-building signal penetration to a densely populated urban environment or provide coverage to remote terrains. Deploying LTE relays would be a more cost effective strategy in these cases.

An LTE relay is a device that can replicate signals or forward traffic between its two wireless interfaces under the LTE air interface specification. An LTE relay serves the traffic of its LTE User Equipments (UEs) on one wireless interface and is served by a wired Donor eNBs (DeNBs) on the other wireless interface.

Extensive studies have been conducted on the type of essential functionalities an LTE relay should possess so that the relay deployment can provide benefits in capacity enhancement, coverage extension, and QoS differentiation.

A simple RF repeater can offer coverage extension for eNBs by amplifying and forwarding the received waveform. However, such a repeater cannot distinguish desired signals from interference and noise, and amplifies both. As a result, repeaters typically lower the network capacity.

An advanced repeater can perform some base-band processing and amplified only a portion of the received waveforms thus can provide a better SINR for UEs under its coverage. However, advanced repeaters are unable to provide capacity gains from sophisticated mechanisms, such as power control, beam-forming, and adaptive modulating/coding, since these mechanisms require control channel coordination between the transmitter and receiver that repeaters cannot perform.

MAC centric relays (L2 relays) that decode and forward UE traffic without having the full eNB functionality can capitalize from mechanisms, such as beam-forming and adaptive modulating and coding. However, L2 relays cannot provide traffic session level QoS differentiation to its UE traffic, since the knowledge of Data Radio Bearers (DRBs) and the associated QoS requirements are maintained by the Radio Resource Control (RRC) [2] protocol above MAC. Handover enhancement and advance interference management techniques are also not applicable to L2 relays, since they would require the knowledge of per UE RLC [3] state and the assistance from upper layer control messaging, respectively.

IP relays (L3 relays) that have the perception of DRBs and the full RRC protocol are ideal to provide full QoS differentiation. They can operate as regular eNBs to perform QoS scheduling, buffer management, flow control, enhanced handover, and advanced interference management with careful considerations of the tradeoff between complexity and performance. Currently, 3GPP has determined to standardize the relay operation based on the L3 relay and drive the development of L3 relay related technologies.

The deployment of L3 Relay Nodes (RNs) faces many open problems, including providing compatibility to LTE Enhanced Packet Core (EPC) and LTE Release 8 UEs, minimizing standardization, development and operational complexity, providing sufficient QoS differentiation, reducing over the air overhead, providing security and more. In this paper, we discuss these problems that face the deployment of LTE relays. We also demonstrate that our proposed solutions to these problems are good candidates under various engineering tradeoffs.

In Section II, we present our proposed relay architecture. In Section III, we discuss various issues of providing Quality of Service over the relay. In Section IV, we discuss the techniques that reduce the over the air communication overhead. In Section V, we discuss the security problems and proposed solutions brought by the relay deployment. Conclusions are given in Section VI.

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II. RELAY ARCHITECTURE

The relay architecture design needs to consider the tradeoff of major requirements, such as 1) early deployment, 2) system complexity, 3) and traffic performance.

The requirement of an early relay deployment indicates that the proposed relay architecture should be compatible with the current LTE EPC architecture. In the current EPC architecture, each data radio bearer between a UE and its serving eNB has a corresponding Enhanced Packet Switched (EPS) bearer between the serving eNB and the UE's Serving Gateway (SGW), so that the core network can directly manage data radio bearers. The requirement of an early relay deployment also indicates that the relay architecture design should impose fewer changes in the LTE protocols and should leave fewer essential mechanisms to design. At the same time, the design should be easily extensible so that future architecture optimizations can be introduced with backward compatibility.

A natural design of the relay architecture to fulfill the purpose of an early deployment is to make the relay deployment transparent to both the UE and the core network. This means that the relay needs to have all functionalities of a regular eNB. The control plane and user plane protocols of S1 and X2 interfaces [4] should terminate towards the relay from the core network. In this manner, the relay manages its UE radio bearers (Uu radio bearers) with corresponding EPS bearers. Subsequently, the relay acts as an eNB to its UEs. At the same time, the DeNB can manage its relay radio bearers (Un radio bearers) with their corresponding EPS bearers. Subsequently, the relay acts as a UE to its DeNB. Uu bearers would be aggregated into Un bearers. In this manner, the presence of the relay becomes transparent to both the UE and the core network, so that no new protocols need to be introduced for a basic functioning relay deployment. We identify this simple relay architecture as relay architecture Alternative 1. The user plane aspect and protocol stack of Relay architecture Alternative 1 are illustrated in Fig. 1 and Fig. 2, respectively.

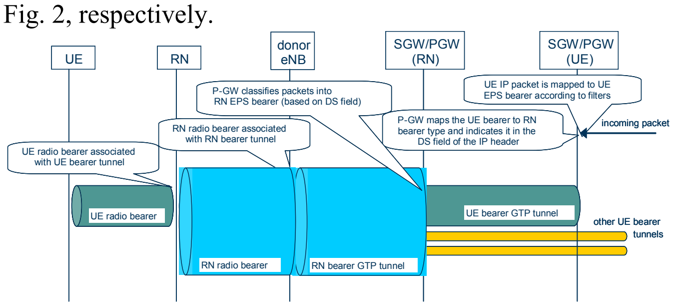

Fig. 1 U-plane Aspect of Alternative 1

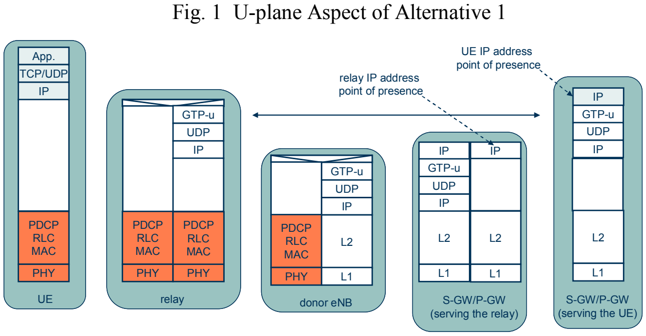

Fig. 2 U-plane Protocol Stack of Alternative 1

Simplicity in architecture often outweighs performance enhancement. Especially, at early deployment, the number of relays and the number of their served UEs are limited. The proposed simple relay architecture is sufficient to provide coverage extension with insignificant amount of traffic overhead being induced by its architectural redundancy.

Alternative 1 has performance disadvantages induced by the fact that the DeNB does not have the visibility of Uu bearers tunneled inside its Un bearers. To remove these disadvantages, a RN's S/P-GW and a RN GW can be deployed inside a DeNB. The embedded RN's S/P-GW eliminates the packet delay between the DeNB and the RN's S/P-GW as well as the operational complexity involved in GTP tunneling. The3 embedded RN GW has the functionalities of a HeNB GW. S1 and X2 interfaces coming from the core network terminate and get their information parsed at the RN GW, while S1 and X2 coming from the relay also terminate and get their information parsed at the DeNB. Subsequently, the DeNB will have per UE flow visibility. We identify this architecture as relay architecture Alternative 2. The user plane aspect and the protocol stack of relay architecture Alternative 2 are illustrated in Fig. 3 and Fig. 4, respectively.

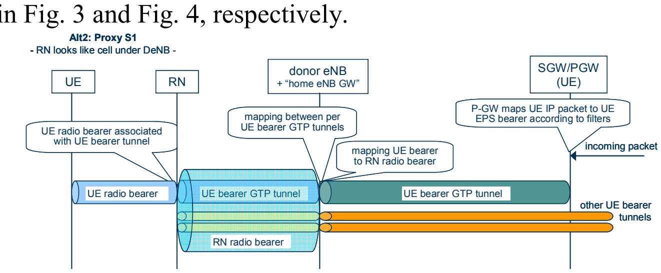

Fig. 3 U-plane Aspect of Alternative 2

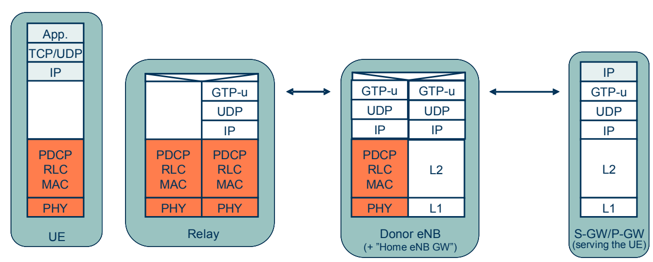

Fig. 4 U-plane Protocol Stack of Alternative 2

With relay architecture Alternative 2, many performance enhancement techniques can be deployed including that:

- The DeNB can exercise call admission control on a per UE flow basis, so that UE flows that have the same QoS and call admission control priority do not have to be rejected or admitted altogether.
- The DeNB can exercise buffer management and scheduling mechanisms on per UE flow basis, such as Random Early Detection (RED), so that fairness can be introduced when the DeNB is experiencing congestion.
- The DeNB can intercept control messages on a per UE flow basis, so that packets do not have to be forwarded back and forth over the relay's backhaul link during handover.
- The DeNB can act as a proxy with a RN GW, S1 interfaces between MMEs and the relay eNBs and the X2 interfaces among relay eNBs and regular eNBs can be

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scaled down to S1 and X2 interfaces originated or terminated at the DeNB.

As the traffic performance is enhanced along with the proposed evolution path from relay architecture Alternative 1 to Alternative 2, the complexity also increases. The proposed architecture also provides a viable way for designing protocols with the backward compatibility.

III. Quality of Service over the Relay

A. Bearer Mapping

The proposed relay architecture imposes flow aggregation of UE bearers into RN bearers over the Un interface. The bear mappings between the UE bearer and the RN bearer under Alternative 2 are illustrated for comparison in Fig. 5.

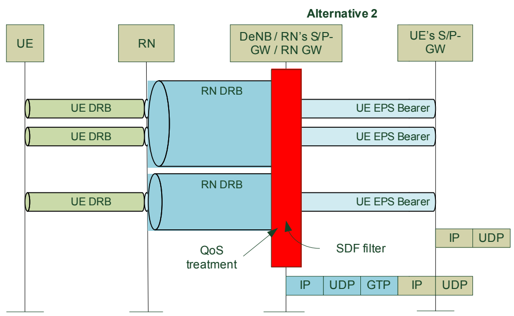

Fig. 5 Bearer Mapping under Alternative 2

In this section, we propose per QoS class flow aggregation with per UE flow buffer management at the DeNB. We show that the proposed bearer mapping can provide sufficient QoS treatment to each UE flow.

The advantage of having per QoS class granularity for RN DRBs is that the bearer management is scalable as the number of RN DRBs does not increase with the number of UE bearers. This significantly reduces system operational complexity. On the other hand, aggregating packets of multiple UE flows that have a common QoS requirement into one RN bearer can induce the Head Of Line (HOL) blocking problem and the TCP window competition problem.

The HOL problem occurs when failed transmissions of one UE flow's packets prevent the successfully transmitted packets of other UE flows to be delivered beyond RLC, since they are aggregated into a common RLC and packets in the common RLC have to be delivered in sequence for the aggregate flow. However, the probability of a transmission failure after HARQ retransmissions is quite low (about 0.1%), which renders HOL blocking with negligible throughput or delay degradation. This is shown in our simulation results using increasingly intense traffic loading over UDP. The results are shown in Table 1 and Table 2. The simulations are conducted using the NS-2 simulator [5] and Qualcomm's proprietary simulator.

Table 1: Performance of 20 UE Flow Aggregations under Poisson Arrival UDP Traffic

Traffic Loading	Performance Metric	Aggregate RLC	Separate RLC
0.25Mbps per flow 5.0Mbps in total	Throughput	4.8Mbps	4.8Mbps
	Service delay	60ms	64ms
0.3Mbps per flow 6.0Mbps in total	Throughput	5.6Mbps	5.6Mbps
	Service delay	200ms	220ms
0.35Mbps per flow 7.0Mbps in total	Throughput	6.4Mbps	6.3Mbps
	Service delay	371ms	384ms
0.4Mbps per flow 8.0Mbps in total	Throughput	7.0Mbps	7.0Mbps
	Service delay	530ms	539ms

Table 2: Performance of 100 UE Flow Aggregations under Poisson Arrival UDP Traffic

Traffic Loading	Performance Metric	Aggregate RLC	Separate RLC
0.05Mbps per flow 5.0Mbps in total	Throughput	5.0Mbps	5.0Mbps
	Service delay	5.9ms	4.3ms
0.06Mbps per flow 6.0Mbps in total	Throughput	5.9Mbps	5.9Mbps
	Service delay	6.8ms	5.8ms
0.07Mbps per flow 7.0Mbps in total	Throughput	6.5Mbps	6.5Mbps
	Service delay	75ms	68ms
0.08Mbps per flow 8.0Mbps in total	Throughput	6.7Mbps	6.7Mbps
	Service delay	619ms	583ms

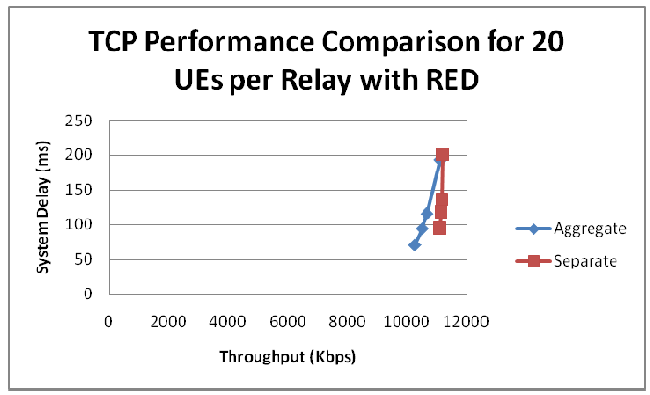

Fig. 6 TCP performance comparison of Aggregate RLC and Separate RLC over the Backhaul Link

In Table 1 and Table 2, the throughput delay results are shown with traffic loading ranging from medium to overload for aggregating 20 UE flows and aggregating 100 UE flows into a single RN flow, respectively. The results show that, aggregating 20 UE flows virtually has no impact on the through and delay performance, while separate RLC approach has slightly lower performance due to higher control overhead induced by separate RLCs. Even with 100 UE flows being aggregated into a single RN flow, the throughput delay performance degradation is not that significant. We note that, the delay throughput curve of 100 UE flows is much steeper and close to the theoretical M/M/1 delay throughput curve compared to that of the 20 UE flows case due to the law of large number effect.

The TCP window competition can occur by aggregating multiple TCP flows into a single RLC. Under separate RLC, the packets of different TCP sessions can be multiplexed at the MAC level with a finer granularity, while under the aggregate RLC, chunk of packets sent out by an opened up TCP window session can get queued in front of a less opened up TCP window session. This causes more frequent TCP window reset under aggregate RLC than separate RLC when congestion occurs. Furthermore, the RED buffer management under separate RLC can drop packets of TCP sessions that have a larger RLC queue size, which gives a more fair TCP feedback to individual TCP sessions. Simulation shows that aggregate

RLC and separate RLC renders similar performance for FTP over TCP, with separate RLC performing slightly better on the delay throughput curve. This is illustrated in Fig. 6.

We propose to deploy one-to-many mapping between RN bearer and the UE bearers on per QoS class basis.

B. Semi-Persistent Scheduling Extension

The Semi-Persistent Scheduling (SPS) [6] operation in LTE Release 8 is designed to support a single VoIP connection between a UE and its DeNB. The SPS operation provides periodic fixed size grants for an uplink VoIP connection to match its packet arrival pattern. In this manner, VoIP packets can be served with less signaling overhead and little delay. The SPS operation can also switch the periodic grants on and off to match the VoIP activity's on and off intervals, so that more VoIP connections can be accommodated by an eNB.

When the same SPS operation is employed on the Un interface to serve multiple uplink VoIP connections, such an operation either allocates an excessive amount of uplink grants that cannot be used efficiently or allocates an inadequate amount of uplink grants that may deliver the VoIP packets delivery or creating higher signaling overhead to accommodate the backlogged VoIP packets. We propose an SPS extension to address the above conflicting interests.

The proposed extended SPS operation allows the RN to specify the number of VoIP connections served through the backhaul link under architecture Alternative. Under architecture Alternative 2, this information is available to the DeNB since the DeNB has visibility of UE bearers. This information can be used as a basic parameter for setting or adjusting uplink grants. Especially, the probability distribution of the number of VoIP connections that are in their talk spurts can be calculated based on this information.

A fixed periodic uplink grant size can be selected between VoIP call arrivals and departures. The selection can be based on the tradeoff between the excessive uplink grants and the overhead and delay incurred by signaling VoIP packets arrivals. The extended SPS operation also allows the RN to request for different persistent uplink grant sizes in between VoIP call arrivals and departures. In this manner, the more frequently adjusted uplink grants can better fit the talk spurts changes. However, adjusting SPS uplink grant sizes too frequently based on the talk spurts changes does not serve the purpose of reducing signaling overhead.

To reduce the frequency of SPS uplink grant size adjustment, we use the following property of superimposed VoIP calls. Assume the number of VoIP calls to be N. Assume the voice activity factor to be ρ. When the number of VoIP calls that are in talk spurts at time t, $n(t) > N\rho$, it is less likely to have $n(t+\Delta t) > n(t)$ and more likely to have $n(t+\Delta t) < n(t)$ for a small Δt. Hence, it is less beneficial to request for a persistent uplink grant size larger than the current superimposed talk spurt size for the next Δt interval. On the contrary, when $n(t) < N\rho$, it is more likely in the next Δt interval that a larger uplink grant size is needed for accommodating incoming talking spurts. This is illustrated in Fig. 7 and Fig. 8.

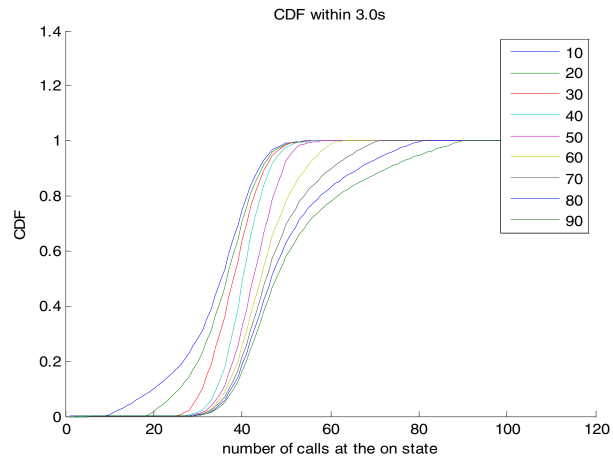

Fig. 7 CDF of the Average Number of VoIP Calls that are in Talk Spurts during the next 3.0s Time Interval

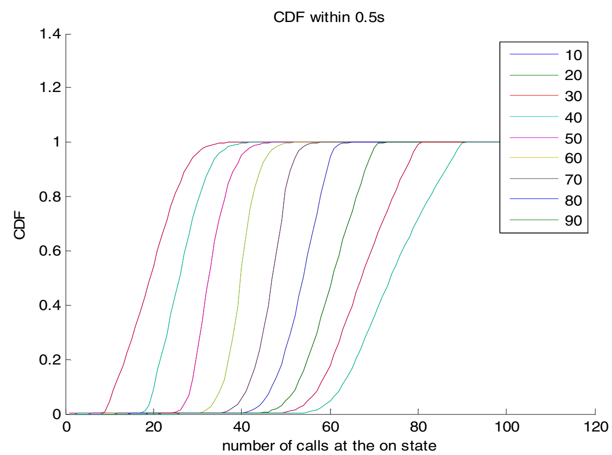

Fig. 8 CDF of the Average Number of VoIP Calls that are in Talk Spurts during the next 0.5s Time Interval

Assume 100 VoIP connections being served by the extended SPS. In Fig. 7 and Fig. 8, we show the CDFs of the average number of VoIP calls that are in talk spurts during a time interval for different initial number of VoIP calls that are seen in their talk spurts. The traffic model of VoIP is in accordance to G.711 [7] and G.792 [8]. As we can see from Fig. 8, when the initial number of VoIP calls seen in their talk spurts is equal to 10, we need to set the uplink grant size larger than 20 VoIP call talk spurts in order to have an 80% probability to accommodate all talk spurts at any moment in the next 0.5 seconds. When the initial number of VoIP calls seen in their talk spurts is equal to 90, we only need to set the uplink grant size to accommodate less than 85 VoIP call talk spurts in order to have an 80% probability to accommodate all talk spurts at any moment for the net 0.5 seconds.

With these CDF figures or their corresponding equations, we can calculate 1) the wasted uplink grant size, 2) the signaling overhead for backlogged VoIP packets, 3) the number of packets that experience extra delay and 4) the signaling overhead for adjusting the SPS grant size. The persistent grant size for the next Δt interval and the Δt interval can be selected based on an arbitrary utility function.

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C. Best Effort Scheduler Extension

The scheduler for best effort traffic at the DeNB does not take into consideration the QoS requirement of individual flows. Instead, the scheduler adopts the principle of proportional fairness [9] and effectively intends to give equal air time for each flow that is visible to the DeNB. The air time is distributed in a manner that flows with better instant channel conditions or lower historic throughputs are scheduled with higher priority. Mathematically, the scheduler strives to achieve the following equilibrium for al flows that have active traffic: $\frac{r_i}{R_i} = \frac{r_j}{R_j}, \forall i, j$, where r_i denotes the instantaneous data rate of flow i; R_i represents the historic throughput achieved for flow i.

Under relay architecture Alterative 1 and 3, the DeNB does not have visibility of UE flows under RN. Hence, regardless how many UE flows are active under an RN, the DeNB will strive to give the RN equal air time compared to a UE that is being directly connected to the DeNB. Hence, we have:

$$\frac{r_{UE_i}}{R_{UE_i}} = \frac{r_{UE_j}}{R_{UE_j}} = \frac{r_{RN_k}}{R_{RN_k}} = \frac{r_{RN_l}}{R_{RN_l}}, \forall i, j, k, l$$

Assume the access channel conditions of UEs under the RN are the same for RN_k, and there are N_{RN_k} number of UEs under RN_k. We will have $R_{RN_k} = N_{RN_k} R_{UE_k^m}$, where UE_k^m is the m-th UE under RN_k. Hence, we have:

$$\frac{r_{UE_i}}{R_{UE_i}} = \frac{r_{UE_j}}{R_{UE_j}} = \frac{r_{RN_k}}{N_{RN_k} R_{UE_k^m}} = \frac{r_{RN_l}}{N_{RN_l} R_{UE_k^n}}, \forall i, j, k, l, m, n$$

which is not the scheduling result the DeNB intends to have. To achieve the original result of proportional fairness from the DeNB point of view, we need to have:

$$\frac{r_{UE_i}}{R_{UE_i}} = \frac{r_{UE_j}}{R_{UE_j}} = \frac{r_{RN_k}}{R_{RN_k}/N_{RN_k}} = \frac{r_{RN_l}}{R_{RN_l}/N_{RN_l}}, \forall i, j, k, l$$

We thus conclude that the DeNB needs to know the number of active UE flows served under RN_k, N_{RN_k}, and then discount the historic throughput of RN_k by N_{RN_k} in order to achieve proportional fairness for best effort traffic.

Under relay architecture Alternative 1, the RN can report to the DeNB the number of active downlink and uplink UE flows under itself by inserting this information in the Buffer Status Report (BSR) messages the RN sends to the DeNB in a continuous or periodic manner. Under relay architecture Alternative 2, the DeNB knows the number of active downlink flows under its RN by monitoring the queues of individual UE bearers. For the number of active uplink flows under its RN, the DeNB can either obtain such information through the BSR sent by the RN or monitor the uplink packet.

IV. Over the Air Efficiency Enhancement

The proposed relay architecture alternatives result in the presence of GTP tunnels of EPS UE bearers between the RN and the UE's S/P-GW. Hence, the outer IP/UDP/GTP header will be in principle transmitted over the Un interface. The composition of the PDCP's payload over the Un interface is illustrated in Fig. 9.

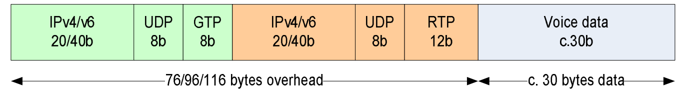

Fig. 9 VoIP Data Protocol Stack over the Un interface

This overhead introduces over the air inefficiency for relay deployment. There are several ways to compress the outer IP/UDP/GTP header and the inner IP/UDP/RTP header.

One header compression scheme can be compressing the inner IP/UDP/RTP header and the outer IP/UDP using ROHC header compression [10] independently. In this manner, no additional ROHC header profile needs to be developed in IETF, which results in the least amount of standardization work within 3GPP and IETF. However, the overhead compressed by using this scheme is less significant.

Another header compression scheme can be compressing both the outer IP/UDP/GTP header and the inner IP/UDP/RTP header, either independently by using a new ROHC header profile for the outer IP/UDP/GTP header, or jointly by using a new ROHC header compression for every combination of outer header and inner header. This scheme results in a better header compression rate, however requires more work in IETF and more coordination between IETF and 3GPP.

By examining the characteristics of the outer IP/UDP/GTP header and the Un air interface, we noticed that the information carried in the outer IP/UDP header are not essential and could be recreated arbitrarily by the RN without the need of dedicated signaling. Hence, the outer IP/UDP header can be simply stripped. Furthermore, the outer IP/UDP/GTP header can be replaced by a 2 bytes Context ID header so that the TEID within the GTP header can be recreated at the RN while the outer IP/UDP header is stripped off. This compression scheme results in a maximum header compression over the Un interface. The only additional work is to standardize the signaling process in 3GPP that creates such Context ID for the outer header stripping. The result of the header stripping is shown in Fig. 10.

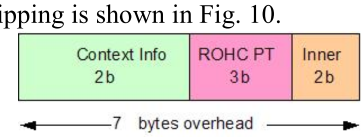

Fig. 10 Header Compression and Stripping under the Un Interface

The signaling process that creates such Context ID can be driven by the RN. The proposed signaling process for downlink and uplink UE EPS bearer over the Un interface is shown in Fig. 11.

In both downlink and uplink direction, the RN signals the DeNB about the expected content of the IP/UDP/GTP header and its associated Context ID (CID) to it. The signaling method reduces the DeNB processing complexity because of the following. There are many IP packets arriving at the DeNB. For the packets that match the context, the DeNB will compress the packets. For those that do not match, the DeNB does not need to do anything further other than forwarding the packets. This is in contrast to other compression scheme that is driven by the DeNB, where if the packet does not match any existing CID's, the DeNB will need to check and decide whether the packet needs to be compressed.

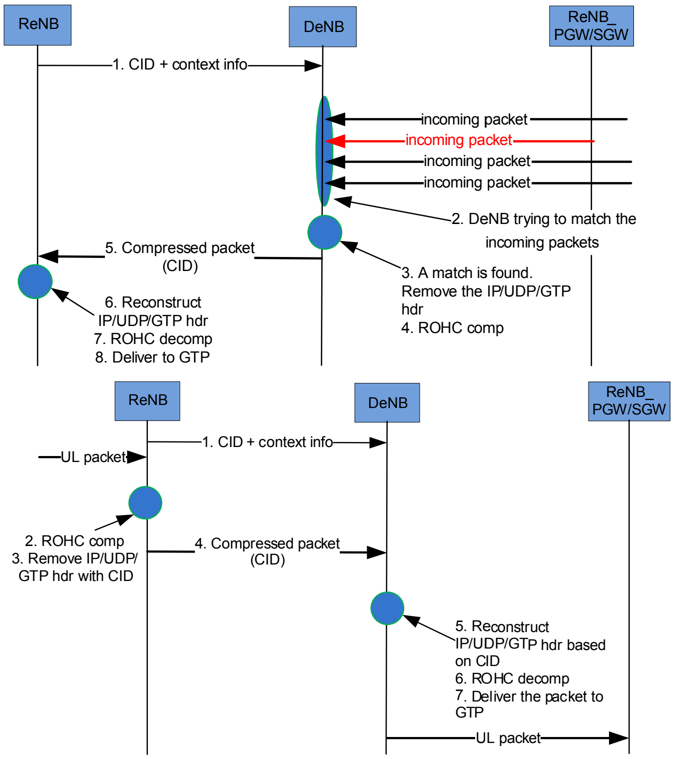

Fig. 11 Signaling for Un Interface Header Compression of Downlink and Uplink UE EPS Bearers

V. SECURITY

By introducing a relay between the UE and DeNB, new security issues are induced. Two security problems easily arise under the existing security procedures in the relay architecture. One problem is that the UE bearer signalling messages are sent over Un interface data bearers without integrity protection. This problem can be solved by providing designated Un interface data bearers for UE signalling messages and turned on the IPSec [11] security integrity protection option for data transmissions over these special data bearers.

The other one is the rogue RN problem, where an attacker removes the UICC from the real RN and inserts it into a fake RN. If the authentication of the UICC with the network is only based on the AKA of the USIM, when the UICC works with a fake RN, any alternation of user plane data committed by the RN will not be detected by the network. This problem is illustrated in Fig. 12.

We propose to solve this security problem by using IPSec over RN's S1 and X2 interface with device authentication. Specifically, the RN should have a hardware key. Upon the establishment of IPSec, the hardware key should be used for device authentication. In this manner, IPSec on S1 and X2 interfaces would stop the keys that relate to the UE being sent to a false RN. IPSec is then used over all signaling planes so that the UEs would not be able to attach or be handed over to a rogue RN.

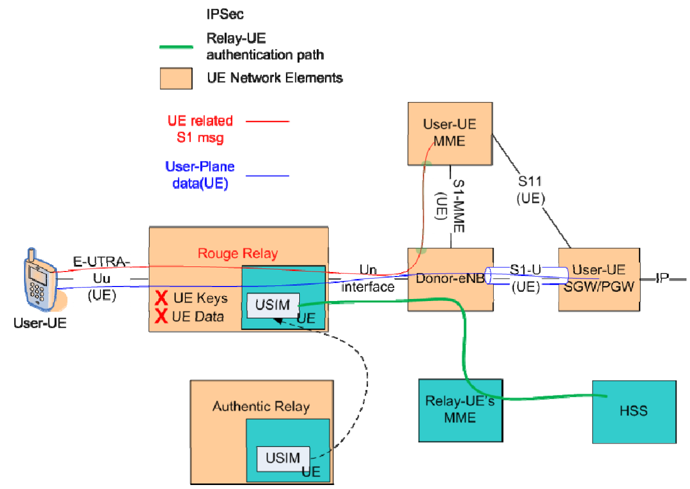

Fig. 12 The Security Problem of a Rogue RN

VI. CONCLUSION

In this paper, we proposed three LTE relay architecture alternatives. We analyzed their trade-offs between complexity and performance. We show that the relay deployment can take an evolution path from relay architecture Alternative 1 to Alternative 2 to realize both an early relay deployment and performance wise optimized relay operations. We then studied and proposed solutions to various upper layer problems induced by the relay deployment, especially in QoS, scheduling, and overhead and security. The proposed solutions have taken scalability and complexity into considerations. We show that the proposed solutions can provide proper QoS treatment to UE traffic served by the relay.

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