Comparison of Various Scheduling Schemes for Voice over LTE Networks

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ABSTRACT

The Long Term Evolution supports high peak data rates (100 Mb/s in the downlink and 50 Mb/s in the uplink), low latency (10ms round-trip delay) in different bandwidths ranging from 1.4MHz up to 20MHz. In mobile broadband networks like LTE, the high performance of the radio network can be realized with proper scheduling of resources for different types of services. The scheduling of resources in the transport network is an area which needs proper attention especially, for real time traffic like VoIP. During periods of congestion, real time services like VoIP can be severely impacted if there is a marginal increase in the end to end delay between VoIP packets or there is a packet loss in the transport network. Therefore, the choice of scheduling strategies plays a key role in guaranteeing good end to end performance for both voice and data services. This paper presents various transport network scheduling strategies for resource allocation and their impact on real time traffic in LTE networks. The study of this paper will be beneficial for understanding basics of LTE networks and scheduling schemes for further deep studies.

Keywords: LTE networks, VoIP, scheduling strategies, QoS.

1. INTRODUCTION

In recent years, mobile communication has evolved rapidly and demand for mobile devices with new and higher quality services is increasing. The existing 3G standard, Universal Mobile Telecommunication System (UMTS), is currently being upgraded with High Speed Packet Access (HSPA) to meet current demands. Later, the 3GPP has worked on the Long Term Evolution (LTE) and intends to surpass the performance of HSPA.

LTE (Long Term Evolution) is an evolution of 4G technology, is becoming popular as the next generation technology supporting high data rates. It is the next step in the evolution of cellular communication data networks which will support mobile broadband services with peak data rates 100 Mbps (downlink) and 50 Mbps (uplink) within 20MHz. Thus LTE will enhance applications such as online gaming and interactive TV. It is expected that in 2014, 80% of broadband users will be mobile broadband subscribers and they will be served by HSPA and LTE networks [6].

The main motivation for LTE has been data services and as a result it has only Packet Switched (PS) domain support. It has a completely packet switched core network architecture unlike its predecessor UMTS which is capable of supporting both the Circuit Switched (CS) as well as Packet Switched (PS) core networks LTE can be deployed in different bandwidths and operates both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD).With TDD the uplink and downlink operate in same frequency band whereas with FDD the uplink and downlink operate in different frequency bands. In the downlink physical layer, LTE uses Orthogonal Frequency-Division Multiple Access (OFDMA) radio technology to meet the LTE requirements for spectrum flexibility and enables cost-efficient solutions for wide carriers with high peak rates. In the uplink, LTE uses a pre-coded version of OFDMA which is Single-Carrier Frequency-Division Multiple Access (SCFDMA) in order to compensate a drawback with normal OFDMA which has a high Peak-to-Average-Power Ratio [13]. Differentiation and scheduling of resources in the transport network plays a key role in guaranteeing good end to end performance for both voice and data services. With packet switched system becoming more popular for carrying data, the Internet Protocol Multimedia Subsystem (IMS) was introduced to ensure QoS for various multimedia services including voice. This alleviates the problem with VoIP calls and ensures that a voice call quality could be as good as the circuit switched based voice call. In traditional wireless networks, real-time services (e.g., voice) are transported over synchronous channels because of their delay sensitivity while data is transported over asynchronous channels because of its burstiness. It has recently been proposed that even real-time services can be efficiently transported over asynchronous channels, such as the time-shared channels supported on the forward link of 3G networks such as 1xEV-DO, 1xEV-DV and HSDPA .In order to provide acceptable user performance for services such as voice over IP (VoIP), while at the same time providing high user capacity, the allocation of resources to users must be carefully managed [9]. In particular, the scheduler that allocates time slots to users must be carefully designed. The rest of paper is organized as follows. Section 2 discusses the LTE network and it's Architecture; Section 3 describes various transport network scheduling strategies for resource allocation. Section 4 discusses the related work about analysis of various scheduling schemes.

2. LTE NETWORKS

LTE encompasses the evolution of the radio access through the E-UTRAN and the non-radio aspects under the term System Architecture Evolution (SAE).The goal of LTE is to increase the capacity and speed of wireless data networks using new DSP (digital signal processing) techniques and modulations that were developed around the turn of the millennium. A further goal is the redesign and simplification of the network architecture to an IP-based system with significantly reduced transfer latency compared to the 3G architecture. The LTE wireless interface is incompatible with 2G and 3G networks, so that it must be operated on a separate wireless spectrum.

2.1 LTE Network Architecture

The LTE network architecture is shown in the Figure 1. The network architecture called the Evolved Packet System (EPS) has a flat IP based architecture and is divided into the Access Network: Evolved Universal Terrestrial Radio Access Network E-UTRAN and Core Network: Evolved Packet Core (EPC) [8].

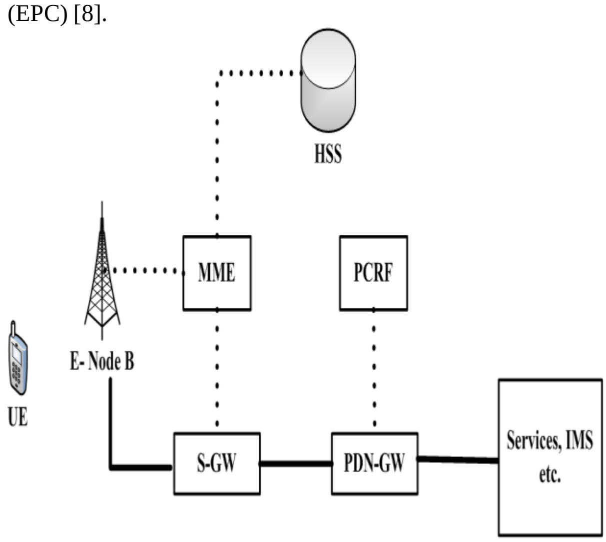

Figure 1 Architecture of LTE Network

The overall architecture consists of five elements which are explained as follows.

- E-UTRAN

The radio network called the E-UTRAN comprises of the E-Node B's that are interconnected to each other over the X2 interface and connected to the core network elements the radio resources for the users in the LTE network. The E-Node B terminates the control plane signaling messages as well as the user plane data with the EPC over the S1 interface.

- EPC-

It is the core network comprising of four elements which are Mobility Management Entity (MME), Serving gateway,

- MME

It is the most important element in the EPC as it terminates the control plane signaling from the user. Some of the functions performed by MME include authentication, mobility management, security and retrieval of subscription information from the HSS.

- Serving gateway

It is responsible for forwarding the user plane packets from the mobile towards the PDN Gateway. It is also responsible for tunneling the user plane IP packets using the GPRS tunneling protocol (GTP) when the user moves across different E Node B's and serves as a mobility anchor for the user plane packets in the LTE network.

- Packet data network (PDN) gateway

It is the end node in the LTE network. It acts as an edge router and routes the user plane IP packets from the mobile nodes to other networks like Internet, IMS etc. It is also responsible for allocation of IP address to the user.

- PCRF

It is responsible for enforcing various operator policies on the network like guar anteed QoS, maximum bit rate provisioned for a user etc. It communicates with the PDN-gateway in enforcing these policies for various users in the LTE network.

- HSS

It is the master database containing all the subscription information of the user along with the subscription for various services that are offered by the operator. It also comprises of the authentication centre which stores all the keys required for ensuring the encryption and integrity of the data in the network.

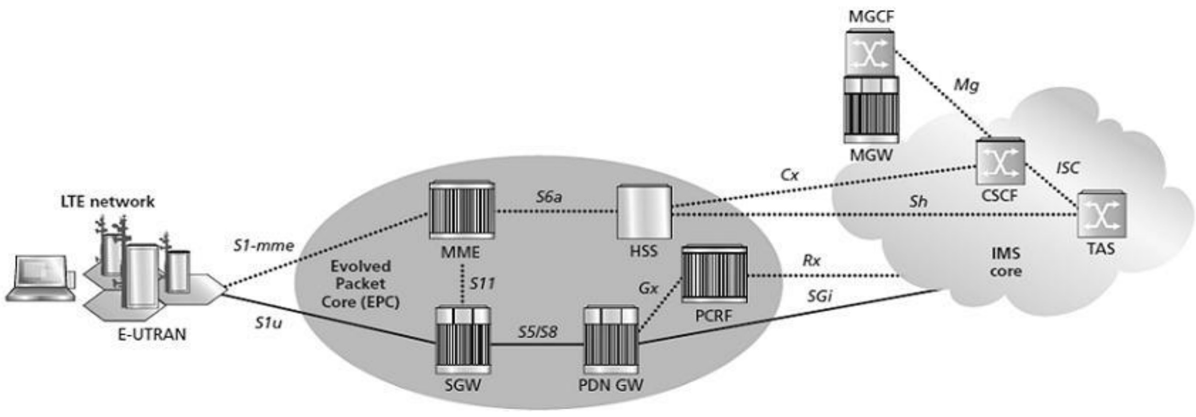

Figure 2 VOICE OVER LTE via IMS [1]

2.2 VOICE OVER LTE

In complete packet switched network, as there is no CS domain the question arises how to provide the voice call over LTE and how the voice call continuity is ensured when the user moves from LTE to 2/3G networks.
Various solutions available in the market today

2.1.1 Voice over LTE via IP Multimedia Subsystem (VoIMS)

It uses IMS call control as defined by 3GPP TS 23.228 for LTE voice-services delivery. IMS provides legacy voice services, such as basic voice origination/termination, calling line identification, and supplementary services, as well as value-added, advanced multimedia services such as video sharing by supporting media additions and subtractions at any time during the call. IMS is a core network architecture that is integrated on top of the LTE network as shown in Figure 2. The IMS network is mainly used to provide all the basic services for voice that are provided by the existing CS networks. In addition, it also provides enhanced multimedia services like video conference, real time gaming etc. The main advantage of using an IMS based solution is that it completely utilizes the LTE architecture rather than relying on the existing CS networks for supporting voice feature. The IMS network is also capable of integrating with the legacy 2G/3G networks and thus can support voice call continuity even when the subscriber moves out of LTE coverage. Hence, the subscriber can experience the same services even when roaming into legacy networks. This solution is being projected as the long term solution as it is capable of providing enhanced services to the LTE network and also supports integration with the existing 2G/3G networks. VoIMS gives operators with wireless and wire line networks the opportunity to offer converged fixed and mobile services, thereby increasing revenue and reducing subscriber churn.

VoIMS is also a good option for MSPs with both GSM/UMTS and CDMA networks because IMS offers convergence between fixed and wireless as well as between different wireless access technologies. VoIMS network implementation requires the deployment of the IMS core — CSCF, Telephony Application Server (TAS), and other components — if not already present in the network, along with all necessary changes to the back-office systems. VoIMS terminals must also support the IMS mobile client. In addition, an IP-SM-GW is required for the support of SMS. An upgrade may also be necessary to the Home Subscriber Server (HSS) to support the presence of the new IP-SM-GW in the network.

3. SCHEDULING METHODS

A scheduler assigns the shared resources (time and frequency) among users terminals. The scheduler controls the allocation of shared time-frequency resources among users at each time instant. The scheduler is located in the base station and assigned uplink and downlink resources. The scheduler determines to which user the shared resources (time and frequencies) for each TTI (Transmission Time Interval) (1ms) should be allocated for reception of DL-SCH transmission. Scheduler performances have the highest impact on the level of service a packet receives. The simple scheduling group includes strict priority and round robin scheduling. A strict priority queuing (PQ) scheduler orders queues by descending priority and serves a queue of a given priority level only if all higher priority queues are empty. Round robin (RR) scheduler, on the other hand, avoids local queue starvation by cycling through the queues one after the other, transmitting one packet before moving on to the next queue. Neither strict priority nor RR schedulers take into account the number of bits transmitted each time a queue is served. A number of scheduling algorithms have been developed to meet this need. Most schedulers can be classified into one of the following categories [10]: frame-based, delay-based and rate-based schedulers. Frame-based schedulers include the weighted round robin (WRR) and hierarchical round robin (HRR) schedulers. Both are non-work conserving schedulers and in these schedulers a node can transmit a packet when the packet is eligible. If no packets are eligible for transmission, the node will become idle even when there are packets in the queue. Deficit round robin (DRR) is a work conserving, frame-based scheduler. In work conserving schedulers, scheduler is idle only when there are no packets waiting to be transmitted. The most typical delay-based scheduler is the earliest deadline first (EDF), in which packets are scheduled in order of their deadlines. GPS (Generalized processor sharing) and WFQ (Weighted fair queuing) schedulers are examples of rate-based schedulers.

3.1 FIFO (First in First Out)

The simplest of all possible scheduling disciplines is the FIFO. In this discipline the scheduler transmits incoming packets in the order they arrive at the output queue, and drops packets that arrive at a full queue. The greatest disadvantage with FIFO is that scheduler cannot differentiate among connections and cannot explicitly serve some connections which may be of more privilege than others.

3.2 Weighted Round-Robin (WRR) Scheduling

It is based on the round-robin and priority scheduling algorithms. All processes in the job queue are given time on the processor in the form of a time slice, thereby eliminating concern for starvation. Also short processes that get re-queued often will not have to wait for longer higher priority processes since all jobs in the queue will be granted time on the processor. This is why the value of the time slice (minimum value), prior to change in relation to weight, is important. This time slice variable in WRR is varied for each individual job according to the weight given to it by the operating system. The time slice will never be below a certain threshold while as the weight increases the time slice increases. A higher weight process is given a longer time slice. This will give critical jobs a longer time on the processor per iteration. WRR will integrate an aging process, alter the time slice for each process according to ―weight‖, and also reorder the process queue according to ―weight‖.

3.3 Priority Queuing

Priority Queuing (PQ) is the basis for a class of queue scheduling algorithms that are designed to provide relatively simple method of supporting differentiated service classes. In classic PQ, packets are first classified by the system and then placed into different priority queues. Packets are scheduled from the head of the given queue only if all queues of higher priority are empty. Within each of the priority queues, packets are scheduled in FIFO order. Some of the PQ benefits are relatively low computational load on the system and setting priorities so that real-time traffic gets priority over applications that do not operate in real time. But, if the volume of higher-priority traffic becomes excessive, lower priority traffic can be dropped as the buffer space allocated to low-priority queues starts to overflow. This could lead to complete resource starvation for lower-priority traffic.

3.4 Strict-priority Scheduling

It is implemented by the special strict-priority scheduler node which is stacked directly above the port. In this algorithm representation of packets are by the scheduler depending and the packets are assigned into different priority queues, the queues are served in accordance to their priority from the highest to the lowest i.e. the Queues which are stacked on top of the strict-priority scheduler node get bandwidth always before other queues. Fair queuing between the queues with same priority level. It configure one queue per interface to have strict-priority(Figure 3) , which causes delay-sensitive traffic, such as voice traffic, to be removed and forwarded with minimum delay.

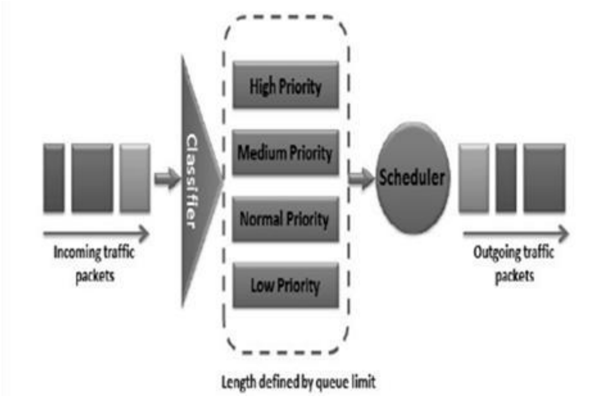

Figure 3 Strict Priority Scheduling

As long as delay sensitive packets are present in the queuing system they will be served first, delay insensitive packets can thus only be transmitted when no delay sensitive traffic is present in the system. Clearly this is the most rigorous way to meet the QoS constraints of delay sensitive traffic. Packets that are queued in a strict-priority queue are removed first and then the packets in other queues are removed including the high-priority queues.

3.5 Weighted Fair Scheduling

In Weighted Fair scheduling defined in [5], the packets are grouped into various queues and each queue is assigned a weight which determines the fraction of the total bandwidth available to the queue. The bandwidth for each queue is based
on the weights and is expresses as

$$BWk = Wk/W*BW$$

It also supports variable-length packets, so that flows with larger packets are not allocated more bandwidth than flows
with smaller packets. The Weighted Fair scheduling assigns the bandwidth for each service based on the weight assigned to each queue and not based on the number of packets. Hence when various types of traffic like VoIP, FTP, and HTTP are flowing in the network, the bandwidth for each service is proportional to its weight and independent of the size of the packet in the queue. The main difference between Weighted Round Robin and Weighted Fair is that the former does packet by packet scheduling in each turn whereas the latter does bit by bit scheduling. Weighted Fair hence has an advantage in the fact that it is aware of the true size of the packets in each queue while performing scheduling whereas Weighted Round Robin is not aware of the same.

4. RELATED WORK

The transport of voice over LTE has a lot of challenges with respect to QoS. In the literature, there are a number of studies which are focused on the optimum scheduling of resources for supporting VoIP service. Most of these studies are focused on the scheduling of resources in the LTE radio network. To the best of my knowledge there are very few studies that have been done on analyzing the impact of scheduling in the LTE transport network. With the need of time many scheduling schemes have been proposed. This section describes literature survey of various scheduling schemes and their performance analysis.

Zoric [17] presented evaluation results of a simulation based study on fairness of different scheduling mechanisms on a real-time traffic streams entering a DiffServ based IP network. It investigated performances of FIFO, PQ, and WFQ schedulers and their impact on voice and video traffic classes in terms of delay and jitter, as well as on data traffic class in terms of throughput of bottleneck link. FIFO scheduler is the worst scheduling choice for voice and video traffic classes, especially with high link utilization. It has been shown that for delay sensitive applications, PQ scheduler has the best performances in terms of delay and jitter. On the other side, using this type of scheduler degrades data traffic, and then is throughput of bottleneck link for this lower-priority class the worst. WFQ scheduler performs better, providing fair amount of bandwidth for data traffic class. Assigning appropriate weights to WFQ scheduler, has significant influence on improving performances of real-time traffic and provides high Quality of Service. In general we can say that WFQ scheduler has better performances among the all compared algorithms, and it is the most suitable for providing QoS for all classes of traffic. When PQ scheduler is used, high priority traffic can easily exhaust traffic with smaller priorities, unless bandwidth limits are configured properly. With proper configuration, WFQ can offer similar type of preferential treatment to high priority traffic, with the advantage of not completely exhausting classes with lower priority treatments.

Strict Priority Queuing (SPQ) and Weighted Round Robin (WRR) are two known disciplines for providing bandwidth guarantees to individual flows. Yue Qian, et.al [15] have applied network calculus to study the Strict Priority Queuing (SPQ) and Weighted Round Robin (WRR) scheduling with respect to the worst-case timing behavior of individual flows in the network. For WRR, proper weights were automatically assigned (using weight allocation algorithm suggested in this paper) for each type of flow to satisfy their delay constraints. On comparing, the service behavior, it has been found that: Firstly, WRR serves traffic in fair manner whereas SPQ is unfair since flows with low priority may be starved. Secondly, WRR is more flexible for QoS provision. WRR enables to balance the allocation of shared network bandwidth to different traffic flows respecting their delay constraints. Thirdly, WRR is capable of providing isolation to individual flows. It is insensitive to other flows' traffic patterns. The per-flow end-to- end delay bound is guaranteed independent of the behavior of other flows.

Ericson and Wanstedt [7] have investigated down-link scheduling performance of MMTel (IMS Multimedia telephony) services for different scheduling algorithms in an HSDPA system. On investigating schedulers over a wide range of traffic mix ratios, it has shown that it is beneficial to have separate schedulers for each media flow. With an MMTel traffic mix of VoIP and FTP flows, the best VoIP capacity was achieved by giving absolute priority to VoIP. However, this led to a relatively poor total cell throughput, i.e. bad FTP throughput. Single schedulers RR and PF perform reasonably well, while Max CQI performs poorly. When delay scheduler for VoIP was used then best performance was achieved and in case of FTP when a proportional fair (PF) scheduler was used then the best performance results were obtained. An extra priority weight for VoIP over FTP (or v.v.) makes it possible to control the wanted total service mix. With both VoIP and real-time video at the same time the best performance was obtained when both services utilize a delay scheduler, this since the VoIP and video flows shall be synchronized. However, different parameter settings are required by the scheduler for the video flow as compared to the VoIP flow to give the best performance.

Siomina and Wanstedt [14] have analysed that that traffic differentiation and service prioritization are particularly crucial when a delay-critical service, e.g., VoIP, is in combination with a delay-insensitive intensive traffic. By prioritizing VoIP, VoIP capacity comparable to that in pure VoIP simulations can be achieved at a cost of a few-percent capacity loss of the second service. Service differentiation and prioritization of delay-critical traffic are important at high loads as well as at any load when multiple services run simultaneously at a user terminal which is a likely scenario in LTE networks. This is especially important when a delay- critical service, e.g., VoIP, is in combination with a delay- insensitive intensive traffic, web surfing or TCP download. It has shown theoretically and by simulations that prioritization of such a service as VoIP typically does not cause large quality degradation of other services due to small VoIP packet sizes but allows more efficient radio resource utilization. It also concluded that in the QoS scenario, more (VoIP) users could be allowed in the system together with the mixed traffic users.

Choi, et.al [3] proposed a Medium Access Control (MAC) layer PRB scheduling algorithm; the key ideas of this scheme are VoIP priority mode and its adaptive duration management. The VoIP priority mode assigns PRBs first to VoIP calls and it is also able to minimize VoIP packet delay and packet loss while the adaptive duration management is able to prevent the overall system performance degradation. The duration of the VoIP priority mode is dynamically adjusted according to VoIP packet drop rate. Their results have shown that, the packet drop rate rises rapidly as the number of VoIP call increases, when the VoIP priority mode is not used. On the contrary, the drop rate remains at low level around 1 % in spite of the increase of VoIP calls in VoIP priority mode.

Dajie et.al [4] presented an effective scheduling scheme known as semi-persistent scheduling for VoIP service in the LTE system. For effectively supporting VoIP service in LTE system the two main challenges are: firstly, tight delay requirement combined with the frequent arrival of small packets of VoIP traffic and secondly, lack of radio resources along with the control channel restriction in LTE system. So it was urgent to design effective scheduling methods to meet the requirements for Quality of Service (QoS) such as packet delay and packet loss rate of VoIP in LTE system . The semi-persistent scheduling combines persistent scheduling for initial transmissions and dynamic scheduling for retransmissions.

This paper shows that high system capacity is supported in semi-persistent scheduling along with guaranteeing the requirements for QoS such as packet loss rate and packet delay of VoIP at the same time. The results of simulation show that in uplink direction high capacity is supported by semi-persistent scheduling and at the same time guaranteeing

VoIP QoS requirements. Furthermore, less control signaling overhead is required in semi persistent scheduling which is very important in a practical system for efficient resources utilization . It also recommended that dynamic scheduling still be the baseline scheduling approach for the VoIP. Only if the signaling load becomes too high, a part of the VoIP users can be scheduled in a semi-persistent fashion.

Zaki, et.al [16] studied the impact of the multi QoS aware MAC scheduling on end user application performance and proposed a QoS aware MAC scheduling algorithm and implemented in the LTE downlink. The proposed scheduler differentiated between the different QoS classes and their requirements. Two different QoS classifications were considered in the paper: Guaranteed Bit Rate (GBR) and non Guaranteed Bit Rate (non-GBR). The goal was to show how to serve VoIP, Video, HTTP and FTP users while satisfying the different QoS requirements of each bearer. The results showed that VoIP bearers survived even when they were mixed with the non- GBR bearers in one MAC QoS class. The reason was the proportional fair nature of the scheduler algorithm, because the VoIP bearers have relatively small data rate which inherently gave them higher priority. The results also showed that since HTTP has a relatively lower data volume to transmit as compared to FTP, it is better not to mix with FTP in one MAC QoS class. Instead, it's better to assign the HTTP on a higher MAC QoS class than FTP.

5. CONCLUSION

As different scheduling mechanisms have different impact on various services under different traffic scenarios. So, the selection of particular scheduling scheme depends upon the parameters that are used to determine the QoS of the various services in the LTE network.

From review of many papers, it has been observed that FIFO and Priority Queuing are not suitable for high speed networks due to their tendency to drop large amount of packets and poor reception of voice data. The bursty nature of WFQ doesn't make it to receive any voice traffic.

This paper would be helpful for readers to learn the basic concepts of LTE networks and scheduling strategies before going into the details of a particular scheduling strategy in transport network, as well as researchers aiming at deepening more specific aspects.

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