# An Analytical Approach to the Efficient Real-Time Events/Services Handling in Converged Network Environment

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**Abstract.** Converged network seamlessly integrates different communications media such as data, voice and multimedia on a single platform. It refers to convergence both types of network and technologies as well as convergence between the different layers of network architecture. In this paper, we examine a priority-based queuing model and perform the mathematical analysis of different media calls processing in converged network environment. We use for this purpose a queuing system model M3/G3/1/NPRP in order to process effectively input jobs/requests (or packets). Tasks within this queuing system get a higher priority if they are handling a real-time event. We present in our paper mathematical results of the expected response and waiting time, and build hypothetical diagrams for the further practical usage in real-time system. A modeling method developed in this paper will be used for the fast configuration and testing of new converged network applications and services.

**Keywords:** Converged Network, M3/G3/1 priority-based queuing model, Non-PRe-emptive Priority (NPRP), Real-time Event, VoIP, Integrated Services.

## 1 Introduction

Networks of today and tomorrow are built on the convergence of voice, multimedia, and data. In this environment, data networks carry voice, video, and data traffic along a managed, secure, and transparent backbone. A converged network affords interoperability among differing communication platforms and allows to have the full range of possibilities that bandwidth allows. This network moves toward a single protocol that can handle the convergence of multiple data types. The marketplace trend is undoubtedly on the way to tighter and better integration of networks because of increased technical performance and lower total IT costs. But, a converged network must be standardized and robust enough in order to handle audio, video, and e-business transactions with the higher requirement degree of security. It has to

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integrate seamlessly different communication types: for the delivering of e-mails and faxes that can be read over the PC or the phone, to allow for the live videoconferencing, and to let users initiate and receive phone calls at the PC [1, 2, 4].

Mainly, information exchanged over the public telecommunication networks has been voice. Present voice communication networks (e.g., Intelligent Networks) utilize digital technology via circuit-switching. The circuit-switching establishes a dedicated path (circuit) between the source and destination. This environment provides fixed bandwidth, short and controlled delay. It provides satisfactory quality of services and does not require a complicated encoding algorithm. Also, in circuit switching, capacity and resources cannot be shared by other users, thereby hindering the system's overall efficiency. In packet-switched network, data is split into packets containing destination identification that are sent and routed independently. It implements store-and-forward switching of discrete data units (packets), and implies statistical multiplexing. This is an ideal environment for non real-time data, where the performance of a best-effort delivery model in terms of throughput is more desirable than delivery of packets within bounded delay and jitter. Crudely sending voice or video data over such a network will lead to poor and even unacceptable quality [3, 5].

To transport voice (e.g., real-time media) over a packet-switched network, it is required a mechanism (e.g., Voice over Internet Protocol - VoIP). The goal of VoIP is to provide the efficiency of a packet-switched network while rivaling the quality of a circuit-switched network. The quality of VoIP does not yet match the quality of a circuit-switched telephone network, but there is an abundance of activity in developing protocols and speech encoders for the implementation of the high quality voice service. The redoubtable problem is that the Internet was designed for data communications; consequently, packets suffer a long and variable delay that decreases voice quality. To overcome this problem, protocols are being developed to provide a certain share of network resources for each voice call through the network. On the whole, many proprietary technologies for VoIP are available, and it is expected that these applications expand as the technologies mature into certified standards - forming a single standard that is an amalgamation of current schemes [4].

In order to overcome mentioned above problems while transporting real-time media in packet-switched networks, we examine in this paper a priority-based queuing system model for the performance of the mathematical analysis of different media calls processing in converged network environment. We assign prioritization rules for different categories of requests (e.g., packets), taking into consideration their importance. Tasks within a queuing system get a higher priority if they are handling a real-time event [4, 6, 14].

We present in our paper theoretical results of the expected response and waiting time in order to build hypothetical diagrams for the further practical usage. To be precise, we give normalized values of main parameters (e.g., arrival rate, service time, time in system). And, when applying real values of $\lambda$ and $\mu$, we have possibility to compare theoretical and practical results (for example, comparison of the theoretical values of time in system $T$, with practical ones), which gives great opportunity to modify and/or adjust real parameters for the better performance of the whole system.