Improving VoIP traffic in LTE

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Telephony has always been the most important of all personal mobile communication services, contributing a major portion of the overall revenue of a telephony operator. The introduction of Long Term Evolution (LTE) has raised the performance bar. LTE has been standardized within the 3rd Generation Partnership Project (3GPP) and is optimized for packet data transfer [1]- [2]. Furthermore, the core network is now solely packet switched, which means that speech is transmitted entirely through Voice over Internet Protocol (VoIP), also known as IP telephony.

The key factors analyzed when assessing network quality are latency, jitter, and packet loss [3]- [4].
Latency (measured in milliseconds [ms]; closer to 0 is better) is how long it takes voice packets to go from their original source to their intended destination. For VoIP calls, this round trip process should take fewer than 100 milliseconds. If the delay is too long, it will cause deterioration of the quality of the phone call and the quality of VoIP services.
Jitter (measured in milliseconds [ms]; closer to 0 is better) refers to the variance in packet delay. If the network has high jitter, the call quality will be affected in ways that result in issues such as garbling.
Packet loss refers to minimal packets of data being dropped as they travel across the network. Low packet loss helps prevent poor call quality or even dropped calls.
In this paper, the IMT-Advanced Indoor Hotspot scenario was considered in order to evaluate LTE VoIP capacity. Moreover, semi- persistent and PDCCH models are discussed and subsequently used to evaluate the system's performance. The OPNET simulator was used to simulate different scenarios.

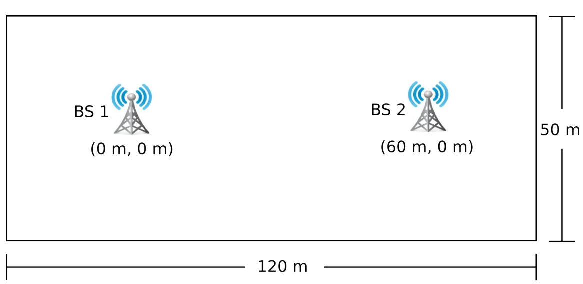

Figure1. IMT-Advanced Indoor Hotspot scenario

According to simulations results, the following conclusions are listed below:
- Indoor hotspot scenario is considered because high user density is mostly observed indoor;
- Control channel constraints restrict the number of schedulable calls in a sub-frame, thus effecting the overall system capacity;
- VoIP capacity is reduced when the control channel's limitations are taken into account.

KEYWORDS

VoIP, latency, jitter, Quality of Service (QoS), Signal-to-Noise Ratio (SNR), packet loss, OPNET

FUTURE WORK

Based on the research and implementation done in this article, some aspects that should be interesting to investigate in the future are:
• Impact of small scale fading
• Identification of users experiencing the worst channel conditions.
Further, the results presented in this article are based on a SIMO system which does not consider the opportunities of MIMO systems. Therefore, an implementation of a MIMO model and an evaluation of VoIP capacity in such a system would provide additional information concerning the performance of the resource scheduling strategies discussed in this paper under realistic control channel constraints.

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