

Serve atOnce Traffica:  
Achieving high quality services  
in Nokia Siemens Networks  
MSC Server System based networks

Nokia Siemens  
Networks



# Quality keeps customers satisfied

Today, end users expect services to be available whenever and wherever they are and with excellent quality. Simply providing connectivity is no longer enough. Although meeting these expectations will allow communications service providers (CSP) to maintain ARPU and customer loyalty, if they want to increase them, they need to exceed expectations – the drive for **quality of service** and **quality of experience** must lie at the heart of every CSP's business.

Providing the required quality is becoming more challenging, particularly with the rapid move towards IP-based networks and the increasing deployment of real-time IP services, such as Voice over IP (VoIP). Mobile voice is becoming more demanding in terms of voice quality and QoS, requiring a real-time monitoring solution that can help solve problems even before the end user knows they exist.

That solution is Nokia Siemens Networks Serve atOnce Traffica.

Traffica helps CSPs to understand how their services are performing by focusing on live traffic and subscriber activity. It presents real-time visibility of end-user activity over the whole network as an overview, at the cell level or even at the level of individual subscribers. Traffica monitors QoS and voice quality delivered to the subscribers and also provides visibility of entire MSC Server System based networks.

# Voice quality and Quality of Service in MSC Server System based networks

From the network point of view, the easiest way to provide high quality service is to ensure the maximum bandwidth is available at all the times to subscribers. However, this would lead to over-investment in the network and significant inefficiency. The challenge is to manage the use of the network's bandwidth and resources so that a required service quality can be provided to all subscribers – all the time – with minimal network investments.

When moving from circuit switched to packet switched technology, mobile CSPs need to implement packet based connectivity between the mobile network elements. In the transition process, CSPs face three main challenges:

- The network's Quality of Service (QoS) scheme must support real-time traffic in addition to non-real-time traffic
- Network resilience should be at least as good as in TDM based systems
- Operational procedures need to be developed to ensure that the packet backbone is operated according to telecom standards.

QoS can vary significantly in IP based networks, particularly with real-time services (such as voice), as IP transmission was originally designed for non-real time services. Therefore, different IP QoS mechanisms have been developed and standardized to guarantee the required service quality in the IP networks. In addition, the convergence of the traditional CS networks and new IP based packet networks, such as IP Multimedia Subsystem (IMS), presents new challenges to end-to-end QoS.

QoS does not create additional bandwidth, but allows the existing bandwidth to be managed and used efficiently to transport the speech packets through the network with predictable delays, jitter and packet loss rates.



## Distributed architecture with MSC Server System

The MSC Server System is based on the 3G Partnership Program's Release 4 specifications (3GPP Rel-4). It separates call control and signaling (control plane) and traffic (user plane) into separate network elements: MSC Server (MSS) and Multimedia Gateway (MGW). This architecture allows a totally new structure for the core network. Call control can be centralized into very large entities, while the actual voice traffic flow can be optimized by locating the MGWs close to the traffic hotspots and the interconnection points closer to other networks.

# See the network from the other side

Serve atOnce Traffica helps CSPs to meet all these challenges. This real-time traffic monitoring and analysis solution for multi-technology, multi-vendor networks reveals the subscribers' view of network performance. It provides a detailed, real-time picture of end-user activity and service use throughout the network. With dynamic load handling to secure network primaries in addition to real-time data flow from the network, it also incorporates embedded high capacity databases, centralized user management and scalable architecture to set the guidelines for future-proof concepts.

The solution comes in two versions, each providing its own specific part of the traffic picture.

**Traffica for MSS** provides real-time visibility of live traffic, subscriber activities and mobility management, either throughout the whole network or down to the cell level. Being connected directly to the MSS provides instant visibility to basic QoS factors such as call setup times and success rates. Mobility management quality indicators such as location updates, handover success rates and handover times are also supported.



It can also provide a wide range of other information:

- How much subscribers are using the services, at what time, from where in the network, and what problems they experience.
- Call completion analysis and mobility management qualities.
- Call details, SMSs and other activities for each subscriber.
- Use of different mobiles and problems per mobile type.
- Activity of user groups, of roamers vs. home subscribers, corporate customers, etc.
- Roaming and traffic destination analysis.

**Traffica for MGW** focuses on true voice quality, QoS, traffic characteristics and stability. Monitoring the user plane from the MGWs gives Traffica visibility of the voice quality and QoS delivered to subscribers. It also allows evaluation of the traffic behavior in the MGWs and forecasting of congestion at particular traffic destinations. Being at the center of the user plane, it also knows the situation in external networks such as radio access, VoIP and the IP backbone.

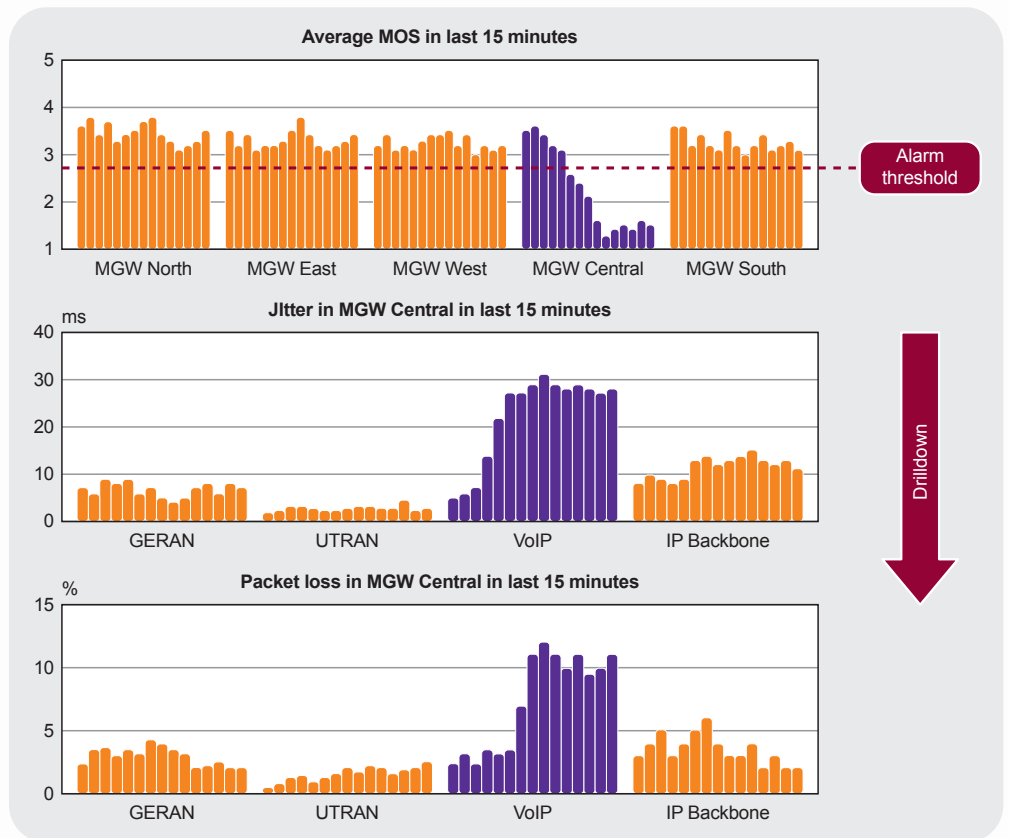
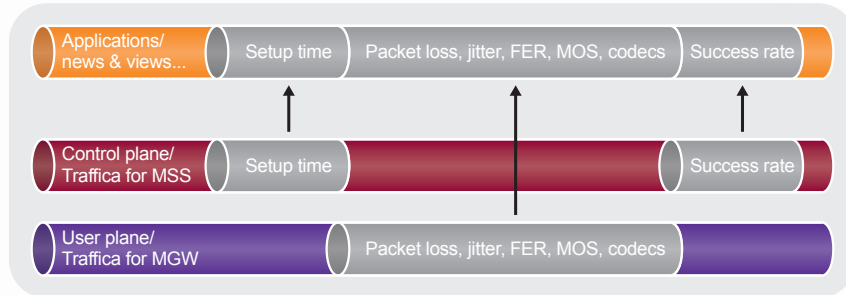
# Best of both worlds

With Traffica, data from the control plane provided by the MSS can be matched to the user plane data provided by the MGW, offering visibility of the quality of experience as the subscriber sees it.

Through its powerful engine for analyzing subscriber event data and using applications such as Traffic News, Traffic Views and Traffica for Customer Care, Traffica can be used at all levels of the CSP's organization. Action to correct problems can be taken immediately and real-time, simplified customer specific data is available to help staff in the front line of customer care to solve problems.

Complete visibility of the entire MSC Server System based network brings many benefits to CSPs' daily tasks and workflows.

- Terminal analysis & service use
- Roaming and traffic destination analysis
- Fast failure detection and congestion forecasting
- Voice quality metrics per subscriber
- Discover trends and threshold for capacity planning
- View of true voice quality as experienced by subscribers
- Evaluate behavior of traffic in MGW and troubleshoot traffic destinations
- IP based route MOS, packet loss and jitter analysis
- Transmission link based MOS and FER analysis
- Voice quality and FER metrics per access interface



## Stop alarms becoming a crisis

The figure above shows how Traffica can help in a situation where a malfunction in the IP backbone causes a sudden drop in VoIP quality. Alarm thresholds can be set to detect abnormalities automatically and to drill down to pinpoint the problem.

In addition, alarms can be forwarded to the network management solution, Nokia Siemens Networks NetAct™, for example, to activate automatic fault correlations.

A number of voice quality factors are available (see Glossary):

- Inter arrival Jitter / Quantile jitter / Peak to Peak Jitter
- IP packet loss ratio (%) / ATM cell loss ratio (%) / Out of order IP packets
- Active speech level information
- Frame erasure ratio (FER)
- MOS (ITU-T P.564) & R-Value
- Noise level information
- Speech activity factor & Double talk
- Echo Path delay / Echo return loss
- Codecs

# Glossary

**Inter arrival Jitter** is a short term jitter calculated over the last packets of the call and gives an overview of the network-level jitter trend caused by the tens or hundreds of terminated calls every second.


**Quantile Jitter** describes IP packet delay variation within a single call, giving an estimate of real jitter encountered by the jitter buffer. It is useful for subscriber specific troubleshooting, i.e. customer care.

**Peak to Peak Jitter** is the maximum difference in delay between two sequential packets within a single call. Nokia Siemens Networks Rel-4 MGW provides a configurable jitter buffer target fill to compensate for the jitter caused by the backbone. The jitter buffer size is a trade-off between the delay and packet loss, i.e. a small jitter buffer decreases the end-to-end delay but increases the packet loss and vice versa.

**Packet Loss** will degrade speech quality, because speech codecs need to compensate for lost packets. If many consecutive packets are dropped, or the packet loss is high enough, speech quality can fall to an unacceptable level. Packet loss can occur during overload situations in the IP network when the routers drop packets randomly depending on their queuing method. Also, over- or underflows in the MGW jitter buffer cause packet loss.

**IP packet loss (%)** describes the ratio of lost IP packets to the sum of all transmitted IP packets, in a population of interest.

**Mean Opinion Score (MOS):** In network quality testing, absolute category rating (ACR), listening-only tests are typically applied to collect subjective responses to the performance of telephone transmission systems. The responses on a five-point categorical scale are averaged to present test results as a MOS score. Each number from 1 to 5 is associated with a definition of audio quality, such as from “poor” to “excellent”. A family of objective algorithms has been standardized in ITU-T for objective evaluation of speech quality. Objective speech quality measurement algorithms estimate subjective speech quality obtained in ACR tests. Furthermore, ITU-T Recommendation P.564 specifies objective speech quality assessment models that predict the impact of observed IP network impairments on the one-way listening quality experienced by the end-user in IP-based telephony applications. In NSN, these models have been expanded to allow MOS evaluation of access interfaces as well as the backbone interfaces.



**Frame Erasure Rate (%)** represents a percentage of received bad/lost speech frames to the number of total speech frames. Traffic is able to monitor both Total FER and Link FER. High FER values may indicate problems in radio access network or in the backbone depending on the measured termination.

**Speech Activity Factor** is the ratio between the active speech time and the total time elapsed during the call. It can be used to aid backbone dimensioning since the average bit rate per call can be estimated from codec information and its speech activity factor. Very high values may also indicate corrupted audio data i.e. faulty channel. Low values may indicate muted calls. Speech Activity Factor is calculated for both the ingress and egress directions.

**Active Speech Level (dBm0)** is a measure of the level (dBm0) of the subscriber's speech. To ensure high understanding and listening comfort, it is important that the received/sent speech level is optimal. Furthermore, the performance of different equipment in the speech path varies depending upon the signal levels. Each active device, such as a speech codec, has a dynamic range over which it functions to specification. Automatic Level Control (ALC) is a useful solution to this problem. It provides an effective, non-obtrusive means of improving the perceived speech quality of a call by automatically optimizing active speech levels. Active speech level information can be used to find the optimum settings for the ALC.

**Noise Level (dB)** measures the level of background noise. High noise levels are very disturbing and can reduce the intelligibility of speech. To overcome these issues and to improve overall speech intelligibility in noisy conditions, background noise level can be reduced in the speech signal with Noise suppression (NS). NS is useful in the network also in cases where there is no or only modest noise reduction in the mobile terminal.

**Doubletalk** is a condition where one participant in a conversation starts talking before the other has finished i.e. both participants are speaking at the same time. High doubletalk ratio can be a result of long end-to-end delays or echo in the connection. Monitoring the occurrence of double talk provides a useful indicator linked to customer perception of quality.

**Echo Path Delay (ms)** is a round-trip delay of the electric echo from PSTN hybrid circuit that MGW echo canceller (EC) detects. It is essential that echo path delay matches the working delay window of MGW echo canceller. By monitoring the echo path delay of each PSTN circuit, the echo canceller can be set to work optimally, or when there are very long PSTN links, the pre-delay setting of the EC can be adjusted to compensate the delays from long links.

**Echo Return Loss (dB)** is a measure of the difference in power between the caller's speech level and the echo that comes back from PSTN hybrid circuit. The higher this value is, the lower the level of the echo compared to the level of the speech, and the better the echo canceller functions.

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