3GPP SRNS Relocation Workshop **Tdoc WHS-99009**
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Requirements for an SRNS Relocation Mechanism

1. Introduction

This paper sets out some general requirements to aid the selection of an SRNS relocation mechanism.

2. Discussion

The range of applications that must be supported is very wide. At one extreme applications can be considered non-real time, whilst at the other extreme applications are very real time in nature. Applications also have very different bandwidth and requirements. Some applications are very bursty whilst others produce continuous streams of data.

Applications should be considered as being on a continuum from non real time to real time. A frame work for SRNS is required which will contain mechanisms to cater for applications any where on the continuum from non real time to real time.
The "framework" must allow for:

- flow classification
- different treatment for different classes of applications
- tuning of parameters (e.g. timeout values, buffer sizes) for different classes of applications

It should not be assumed that all of the classes and parameters will be known at the time of standardization. The framework must be flexible enough to allow new classifications to be added and parameters to be changed (in the field, by individual vendors) without necessitating changes to the standards. "Web time" and all that ;-)

Non real time applications will typically have a zero tolerance of errors. These applications will typically either use an error correction protocol or have in built error protection. Thus some level of data loss is acceptable for these applications. It is suggested that an IP packet loss of between 1-3% on a random basis should not cause problems. However the impact of a 1-3% IP packet loss on a burst basis is not fully understood and may have a far greater impact - this needs further study.

Variation in the RTT (round trip time) can be critical. If the latency of the connection increases by more than one RTT, the impact on the connection is likely to be detrimental.

The critical factor for TCP is the round trip time (RTT). This value is dynamically computed by TCP so that it can adapt itself to run over a variety of links and networks. To estimate the RTT, you need to know:

- the "transmission" delays through the network (related to number of router hops, link speeds along the path, packet length, link layer retransmission delays, etc.)

- the "fixed" delays through the network (i.e. independent of channel speed and packet size, like router header processing or radio link burst negotiation time)

The TCP retransmission timer (RTO) is usually twice the RTT. So when you introduce additional, unexpected delays (like suspend/resume operations) that approach the RTT time, then you start getting into conflicts with the TCP recovery mechanisms.

Currently 3GPP UMTS has set a target end to end delay of 150ms for real time voice, 250ms for real time data with a maximum of 400ms for both. These would indicate a RTT of between 300ms and 800ms far end to far end and back.

Real time applications are generally more error tolerant but cannot cope with an increase in the latency of the connection. The target for these types of application should be virtually no increase in the end to end latency during SRNS re-location.

Duplication of packets and packet re ordering can occur in the fixed network. The effects can be detrimental on protocols and applications. An SRNS relocation solution should not rely on being able to duplicate or re order packets to either far end of the connection. Further work is required to determine at what level duplication & reordering of packets becomes critical.

It can not be assumed that applications will be UMTS aware of specific requirements. UMTS must support fixed network applications that expect to be running on a wired internet.

3. Conclusion
The following requirements have been identified for a solution on SRNS relocation:

- The solution shall support a wide range of applications to cater from non-real time to real time and shall be flexible for future evolution.
- It is proposed that for non-real time applications, packet loss in the range of 1 to 3%, on a random basis, is acceptable. For non-real time applications on a bursty basis, the acceptable packet loss is FFS.
- The solution shall account for TCP mechanisms (e.g. recovery and RTT).
- The impact on latency for real time applications should be minimal.
- The solution shall not rely on duplication and reordering mechanisms that protocols (e.g. TCP) might use to either far end of the connection.