Agenda Item:

Source: InterDigital Comm. Corp.

Title: RACH Capacity Analysis-Packet 1

Document for: Information and Discussion

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Introduction

TSG_R1(99) 614 shows analysis that is based on the assumption that the required demand for RACH capacity is up to 3 accesses per frame.

It is our understanding that the rationale for this required demand, 3 accesses per frame, derives from Document TSGR1#3(99)132 by Siemens.

Referring to the Table in that document, the two critical cases are associated with packet data transmission:

1. Packet 1; substantial amount of packet data (UDD), transmitted in the uplink direction (3 successful access attempts per 10ms frame).

2. Packet 2: Typical "web browsing"; traffic is highly asymmetric, uplink packets transmit HTTP "GET" commands (2.5 successful access attempts per 10ms frame).

In Tdoc TSGR1(99)706, InterDigital demonstrated that there is no need for more than 1.5 accesses per frame for Packet 2. This reference also provided anlalysis for Packet 1, and it was agreed that this analysis should be redone.

This contribution provides an anlysis of the required demand for Packet 1.

Analysis

The analysis that follows uses the terminology and parameters given in Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS (UMTS 30.03 version 3.2.0). The rlevent section, B.1.2.2, Traffic Models, is reproduced and attached below.

Tdoc 132 defines a scenario for Packet 1 with 60% of time slots allocated for uplink. Average Packet size is 500 bytes=4,000bits.

Consistent with this assumption let us assume:
*   2 slots for the downlink control channel
*   8 slots for an uplink shared channel, which supports the packet data
*   1 slot for RACH.

Ignoring the two control channels, the assigment for uplink data is 9/14= 64%

Assume that every uplink shared channel

- is 100% efficient
- uses 1/2 rate coding and SF=8
- exploits multi user detection to support 8 simultaneous messages.

Each time slot supports 244bits per code x 8 codes = 1952 bits

Since 4000/1952 = (approximately) 2,
the capacity of the uplink shared chanel is 1/2 packet per slot. Since there are eight slots assigned for this function per frame, the capacity is 4 packets/frame.

The assumptions above are extreme; 100% efficiency; ability to operate with eight parallel channels with SF=8. Under these assumption, if we design the Uplink Shared Channel to require a RACH transmission per packet, then there could be an argument in favor of supporting up to 4 attempts per frame.

However, it is assumed; i.e. recommended that the Uplink Shared Channel shall be designed to require one RACH attempt per packet call; not one RACH attempt per packet.

According to UMTS 30.03 Annex B, the average number of packets per packet call is 25.

Therefore, the average number of access attempts per frame is 4/25 <0.4

B1.2.2 Traffic models

Real time services

For all real time test services, calls should be generated according to a Poisson process assuming a mean call duration of 120 seconds for speech and circuit switched data services.

For speech, the traffic model should be an on-off model, with activity and silent periods being generated by an exponential distribution. Mean value for active and silence periods are equal to 3 seconds and independent on the up and downlink and both are exponentially distributed.

For circuit switched data services, the traffic model should be a constant bit rate model, with 100 % of activity.

Non-real time services

Figure 1.0 depicts a typical WWW browsing **session**, which consists of a sequence of **packet calls**. We only consider the packets from a source which may be at either end of the link but not simultaneously. The user initiates a packet call when requesting an information entity. During a packet call several **packets** may be generated, which means that the packet call constitutes of a bursty sequence of packets, see [ref 1] and [ref 2]. It is very important to take this phenomenon into account in the traffic model. The burstyness during the packet call is a characteristic feature of packet transmission in the fixed network.
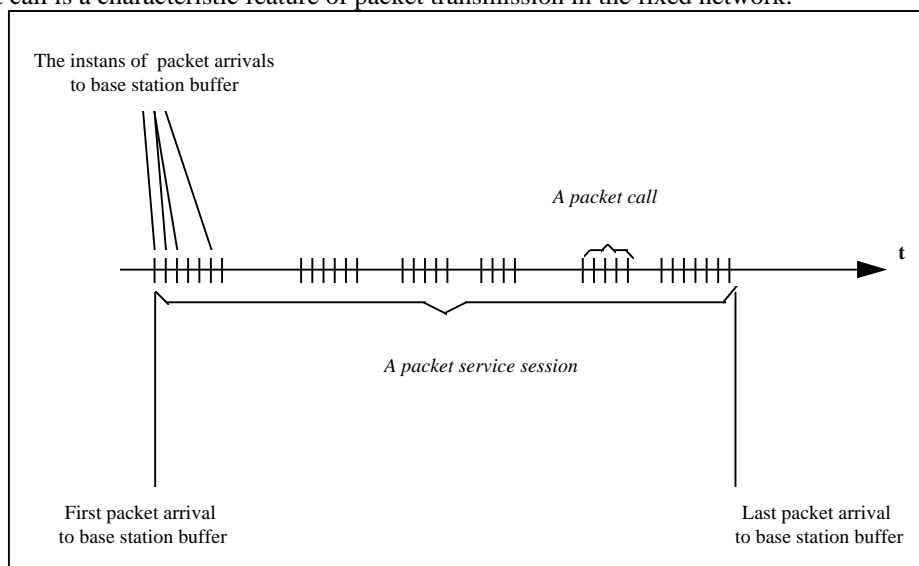

Figure 1.0. Typical characteristic of a packet service session.

A packet service session contains one or several packet calls depending on the application. For example in a WWW browsing session a packet call corresponds the downloading of a WWW document. After the document is entirely arrived to the terminal, the user is consuming certain amount of time for studying the information. This time interval is called **reading time**. It is also possible that the session contains only one packet call. In fact this is the case for a file transfer (FTP). Hence, the following must be modelled in order to catch the typical behaviour described in Figure 1.:

 – Session arrival process

 – Number of packet calls per session, N_{pc}

 – Reading time between packet calls, D_{pc}

– Number of datagrams within a packet call, N_d

– Inter arrival time between datagrams (within a packet call) D_d

– Size of a datagram, S_d

Note that the session length is modelled implicitly by the number of events during the session.
Next it will be described how these six different events are modelled. The geometrical distribution is used (discrete representation of the exponential distribution), since the simulations are using discrete time scale.
Session arrival process: How do session arrive to the system. The arrival of session set-ups to the network is modelled as a Poisson process. For each service there is a separate process. It is important to note that this process for each service only generates the time instants when service calls begin and it has *nothing to do with call termination.*
The number of packet call requests per session, N_{pc}: This is a geometrically distributed random variable with a mean m_{Npc} [packet calls], i.e.,

$$N_{pc} \in Geom(m_{Npc}).$$

The reading time between two consecutive packet call requests in a session, D_{pc}: : This is a geometrically distributed random variable with a mean m_{Dpc} [model time steps], i.e.,

$$D_{pc} \in Geom(m_{Dpc}).$$

Note that the reading time starts when the last packet of the packet call is completely received by the user. The reading time ends when the user makes a request for the next packet call.
The number of packets in a packet call, N_d: *The traffic model should be able to catch the various characteristic features possible in the future UMTS traffic. For this reason different statistical distributions can be used to generate the number of packets.* For example N_d can be geometrically distributed random variable with a mean m_{Nd} [packet], i.e.,

$$N_d \in Geom(m_{Nd}).$$

It must be possible to select the statistical distributions that describes best the traffic case under study should be selected. An extreme case would be that the packet call contains a single large packet.
The time interval between two consecutive packets inside a packet call, D_d: This is a geometrically distributed random variable with a mean m_{Dd} [model time steps], i.e.,

$$D_d \in Geom(m_{Dd}).$$

Naturally, if there are only one packet in a packet call, this is not needed.
Packet size, S_d: The traffic model can use such packet size distribution that suits best for the traffic case under study. Pareto distribution with cut-off is used.
The normal Pareto distribution (without cut-off) is defined by:

$$
\begin{cases}
f_x(x) = \dfrac{a \cdot k^a}{x^{a+1}}, x \geq k \\[2mm]
F_x(x) = 1 - \left(\dfrac{k}{x}\right)^a, x \geq k \\[2mm]
m = \dfrac{ka}{a-1}, a > 1 \\[2mm]
s^2 = \dfrac{k^2 \cdot a}{(a-2) \cdot (a-1)^2}, a > 2
\end{cases}
$$

PacketSize is defined with the following formula:

$$PacketSize = \min(P, \ m),$$

where P is normal Pareto distributed random variable (α=1.1, k=81.5 bytes) and m is maximum allowed packet size, m=66666 bytes. The PDF of the PacketSize becomes:

$$f_n(x) = \begin{cases} \dfrac{a \cdot k^a}{x^{a+1}}, & k \leq x < m \\ b, & x = m \end{cases}$$

where β is the probability that x>m. It can easily be calculated as:

$$b = \int\limits_{m}^{\infty} f_x(x)dx = \left(\frac{k}{m}\right)^a, a > 1$$

Then it can be calculated as:

$$m_n = \int\limits_{-\infty}^{\infty} x f_n(x)dx = \int\limits_{k}^{m-} x \frac{a k^a}{x^{a+1}} dx + m\left(\frac{k}{m}\right)^a = \ldots calculating \ldots = \frac{ak - m\left(\frac{k}{m}\right)^a}{a-1}$$

with the parameters above the average size is:

$$m_n = 480 \, bytes$$

Table 1.1 gives default mean values for the distributions of typical www service. According to the values for α and k in the Pareto distribution, the average packet size μ is 480 bytes. Average requested filesize is m_{Nd} x μ = 25 x 480 bytes ≈ 12 kBytes. The interarrival time is adjusted in order to get different average bit rates at the source level.

Table 1.1: Characteristics of connection-less information types

Packet based information types	Average number of packet calls within a session	Average reading time between packet calls [s]	Average amount of packets within a packet call []	Average interarrival time between packets [s][1]	Parameters for packet size distribution
WWW surfing UDD 8 kbit/s	5	412	25	0.5	k = 81.5 (= 1.1
WWW surfing UDD 32 kbit/s	5	412	25	0.125	k = 81.5 (= 1.1
WWW surfing UDD 64 kbit/s	5	412	25	0.0625	k = 81.5 (= 1.1
WWW surfing UDD 144 kbit/s	5	412	25	0.0277	k = 81.5 (= 1.1
WWW surfing UDD 384 kbit/s	5	412	25	0.0104	k = 81.5 (= 1.1
WWW surfing UDD 2048 kbit/s	5	412	25	0.00195	k = 81.5 α = 1.1

[ref 1] Anderlind Erik and Jens Zander " A Traffic Model for Non-Real-Time Data Users in a Wireless Radio Network" IEEE Communications letters. Vol 1 No. 2 March 1997.
[ref 2] Miltiades E et al. "A multiuser descriptive traffic source model" IEEE Transactions on communications, vol 44 no 10, October 1996.

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[1] The different interarrival times correspond to average bit rates of 8, 32, 64, 144, 384 and 2048 kbit/s.