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Technical Report

3rd Generation Partnership Project;
Technical Specification Group Core Networks and Terminals
Feasibility Study on Bandwidth Savings
at Nb Interface with IP transport
(Release 7)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

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1 Scope

The present document investigates solutions for a simple, optional transport format for the Nb interface and IP transport that allows transporting several RTP/NbFP/codec payload PDUs of different user plane connections within one packet. The solution shall minimise impacts to existing network characteristics, in particular jitter and packet delay. The transport format shall be suitable for any type of payload transported within NbFP. Though primarily intended and possibly optimised for NbFP transport within 3GPP circuit-switched core networks, the multiplexing design shall be generic and future proof by not precluding support of non-NbFP payload types, e.g. standard VoIP bearers or other interfaces such as Iu.

Furthermore, backward compatible signalling extensions required to negotiate the use of this transport option, to set up multiplexing connections, and to assign bearers to these connections are investigated.

The benefits and drawbacks of multiplexing at the Nb interface are investigated.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 29.414: "Core network Nb data transport and transport signalling".

[2] 3GPP TS 29.415 "Core network Nb interface user plane protocols".

[3] 3GPP TS 25.414 "UTRAN Iu interface data transport and transport signalling".

[4] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[5] IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] apply.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AVP Audio Video Profile
BICC Bearer Independent Call Control
CN Core Network
IANA Internet Assigned Numbering Authority
IP Internet Protocol
IPBCP IP Bearer Control Protocol
Iu Interface between the RNS and the core network. It is also considered as a reference point.

IuFP	Iu Framing protocol
MGW	Media GateWay
Nb	Interface between media gateways.
PDU	Protocol Data Unit
RFC	Request For Comment
RTP	Real-Time Transport Protocol
RTCP	Real-Time Transport Control Protocol
SDP	Session Description Protocol
UDP	User Datagram Protocol

4 Requirements

Multiplexing can offer a relatively simple way to increase bandwidth efficiency in an IP network. It should be simple since multiplexing should not necessarily require any additional features from the network and even as little as multiplexing of two packets could save bandwidth significantly.

The destination address and DiffServ class should be the same for all multiplexed packets.

Real-time applications like speech have also strict requirements for e.g. delay and jitter, which have to be taken into consideration. The differentiation of streams in an efficient way is important together with the fact that the multiplexing protocol should not restrict the maximum number of possible connections between any two nodes (e.g. MGWs).

The multiplexing method shall be independent of the protocols beneath IP and it can be used e.g. in an MPLS enabled network as well as in any other IP based network.

VoIP networks using RTP framing should also be supported.

5 Transport Format

5.1 Proposed Format(s)

5.1.1 UDP Port Multiplex Header

This is a new multiplexing method designed for CS traffic transported over IP in a 3GPP UMTS network over Nb-interface between MGWs or over Iu-interface between an RNC and MGW. The method introduces a multiplexing header, which identifies every multiplexed packet. The traffic is assumed to be real-time and the DiffServ class is then the same for all packets.

Multiplexing can be performed for all packets heading to the same IP address and this particular method can be used for all UDP traffic as long as they share the same DiffServ class. The multiplexing is intended to be used only with RTP packets.

RTCP is transported normally by IP/UDP packets.

The UDP port alone is not enough since it does not indicate where the next multiplexed packet starts (AMR or PCM may be used with different lengths). A length indicator (LI) field is thus included in the multiplexing header. The proposed multiplexing header is illustrated in figure 1.

Bits (7 6 5 4 3 2 1 0)	Number of Octets	
Source IP, Dest IP, …	20/40	IP
Source Port, Dest Port=<MUX UDP port>, Length, …	8	UDP
T=0 | Mux ID = (Destination UDP Port of multiplexed PDU) / 2	2	Multiplex Header
Length Indicator (LI) = n	1	
R=0 | Source ID = (Source UDP Port of multiplexed PDU) / 2	2	
Full RTP packet	n	RTP header / RTP NbFP Payload
Multiplex Header	5	Multiplex Header
Full RTP packet	m	RTP header / RTP NbFP Payload
…		

Figure 1: Multiplexing header

The multiplexing header includes

- T bit.
  The field has two possible values, 0 for indicating full packet and 1 for indicating compressed packet. Value 0 shall be used for an uncompressed RTP header, as described in the present sub-clause.

- Mux ID, 15 bits.
  For identification of different user plane connections. The value shall be the same as the UDP destination port of the corresponding non-multiplexed packet divided by two (only even numbered ports are used for RTP sessions).

- Length Indicator (LI), 8 bits.
  Gives the length of the multiplexed RTP packet in bytes (header + payload). Maximum length is 256 bytes (requires padding if last byte is not full). E.g. the payload of AMR 12.2 is only 31 bytes but for future use 8-bit LI may be useful (combined payload of four 5 ms PCM samples resulting in 160 bytes has been proposed).
  LI gives the information where the next multiplexed packet starts.

- R bit, reserved for future extensions. Shall be set to 0 by the sending entity and be ignored by the receiving entity.

- Source ID, 15 bits.
  For identification of the different connections. Value is the same as the source UDP port of a non-multiplexed packet divided by two (only even numbered ports are used for RTP sessions). It is transferred to permit the receiving node to optionally detect and filter illegitimate packets (e.g. packets received from the peer termination precedingly associated to the receiving termination).

The multiplexing can be performed either with common IP/UDP/RTP or IP/UDP header. For voice traffic in a 3GPP network the RTP information is essential and it is thus suggested that entire RTP frames are multiplexed and they together share a common IP/UDP header in Nb- and Iu-interfaces (Figure2. If the packets shared a common IP/UDP/RTP header the bandwidth savings would naturally be greater and it could be used in some special cases where individual RTP information is not needed.

The multiplexing method does not limit the number of packets being multiplexed and it is thus the data link layer protocol that defines the maximum frame size. E.g. an IP datagram has a maximum length of 65 535 bytes and Ethernet 1 518 bytes. In order to avoid additional delay in the network the packets should not be delayed more than 1 ms to 2 ms, which also effectively limits the number of multiplexed packets and makes the multiplexing-jitter low. The time frame should still be enough to gather several packets.

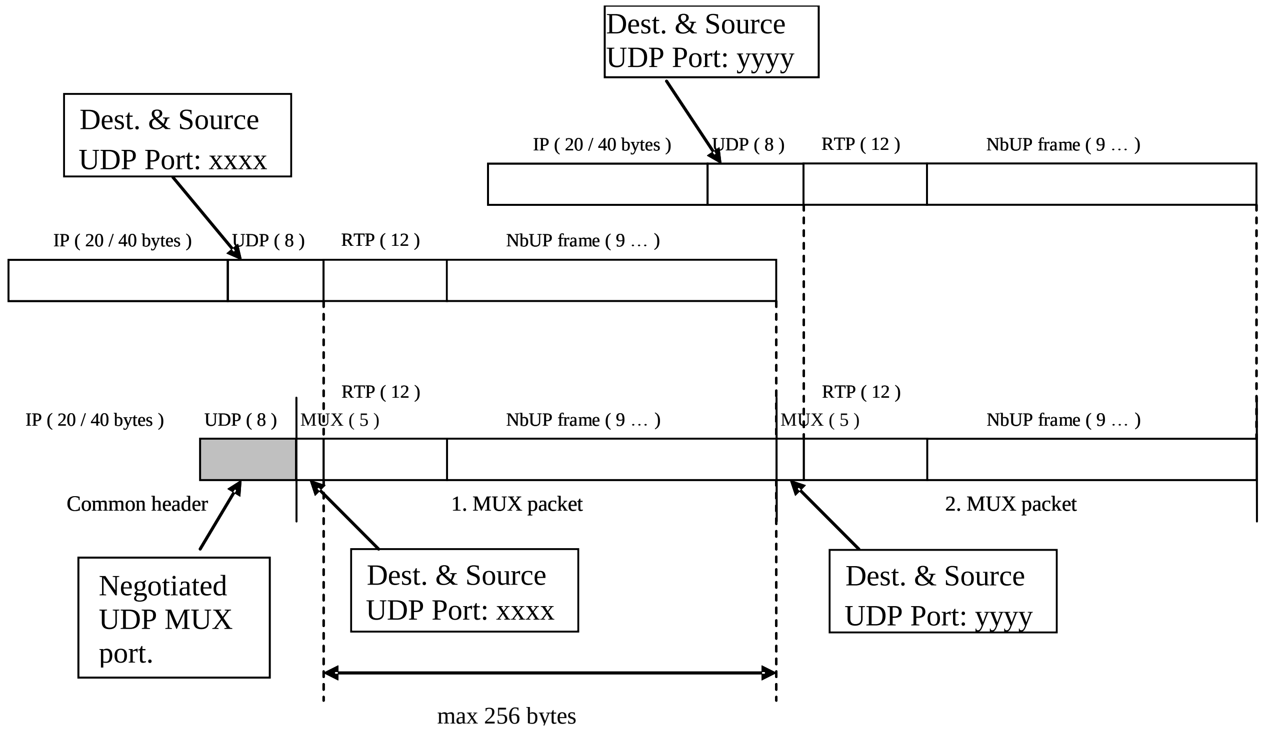

Figure 2: Example of multiplexed packet with two RTP frames

5.1.2 RTP Header Reduction

5.1.2.1 RTP Header Reduction for NbFP transport

In 3GPP NbFP there is only one Payload Type and so this sub-clause applies specifically for NbFP transport within 3GPP circuit-switched core networks.

For NbFP transport, to achieve even better bandwidth savings the RTP header can be compressed. This is possible since RTP header includes many static fields that remain unchanged during an RTP session. Compression shall be an optional feature that must be negotiated between nodes. A connection that has negotiated to use RTP header compression sends all packets into the negotiated UDP multiplexing port. The multiplexing header for these packets is illustrated in figure 3.

In compression there is always an initialization phase first where the full header is transferred to receiver and the type field makes it possible to send also non-compressed packets (also sent to the same negotiated UDP port). The full header is stored and it is used in decompression. After initialization only compressed headers are sent unless information changes in the fields that are not sent within compressed header. For each RTP session, at least the first two RTP packets shall be sent with their full RTP header. RTP packets shall also be sent with their full RTP header at least till receipt of a RTCP packet from the peer indicating support of RTP header compression.

NOTE: The receiving MGW shall not consider as an error if the initialization phase first RTP packets with full header are not received.

The proposed multiplexing and compression RTP headers for NbFP transport are illustrated in figure 3.

<table>
<tr><th colspan="8">Bits</th><th rowspan="2">Number of Octets</th><th rowspan="2"></th></tr>
<tr><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th>0</th></tr>
<tr><td colspan="8">Source IP, Dest IP, …</td><td>20/40</td><td>IP</td></tr>
<tr><td colspan="8">Source Port, Dest Port=<MUX UDP port>, Length, …</td><td>8</td><td>UDP</td></tr>
<tr><td>T=1</td><td colspan="7">Mux ID = (Destination UDP Port of multiplexed PDU) / 2</td><td>2</td><td rowspan="3">Multiplex Header</td></tr>
<tr><td colspan="8">Length Indicator (LI) = n+ 3</td><td>1</td></tr>
<tr><td>R</td><td colspan="7">Source ID = (Source UDP Port of multiplexed PDU) / 2</td><td>2</td></tr>
<tr><td colspan="8">Sequence Number (SN)</td><td>1</td><td rowspan="2">Compressed RTP header</td></tr>
<tr><td colspan="8">Timestamp (TS)</td><td>2</td></tr>
<tr><td colspan="8">RTP payload</td><td>n</td><td>RTP NbFP Payload</td></tr>
<tr><td colspan="8">Multiplex Header</td><td>5</td><td>Multiplex Header</td></tr>
<tr><td colspan="8">Compressed RTP header</td><td>3</td><td>Compressed RTP header</td></tr>
<tr><td colspan="8">RTP payload</td><td>m</td><td>RTP NbFP Payload</td></tr>
<tr><td colspan="8">…</td><td></td><td></td></tr>
</table>

Figure 3: Multiplexing and Compressed RTP header for NbFP transport

The multiplexing header shall be used as described in sub-clause 5.1.1. However, the T bit shall be set for a compressed RTP header, as described in the present sub-clause. The Length Indicator gives the length of the multiplexed RTP/NbFP PDU packet in bytes (compressed RTP header + RTP Payload.

The multiplexed RTP/NbFP payload PDU shall be inserted in the IP/UDP packet directly after the corresponding multiplexing header. The multiplexed RTP/NbFP payload PDU shall consist of the compressed RTP header described below followed by the full NbFP header and the NbFP payload, as described in 3GPP TS 29.415 [2]. If the multiplexed RTP/NbFP payload PDU does not end at a byte boundary, its last byte shall be padded with zeros.

The proposed compressed RTP header includes the following two fields that change during a connection and need to be transferred within each packet:

- Sequence number (SN), 8 bits.
  The field changes as the original sequence number (RFC 3550 [4]) but is shortened from 16 bits to 8 bits (256 packets). The least significant byte of the RTP sequence number shall be included.

- Timestamp (TS), 16 bits.
  The TS field changes as the original timestamp (RFC 3550 [4]) but the length is half of the original resulting in modulo of 4 seconds with 16 kHz clock reference. The least significant two bytes of the RTP timestamp shall be included.

NOTE: These RTP fields change during a connection and thus need to be transferred within each packet for NbFP payload. All other RTP fields do not change.

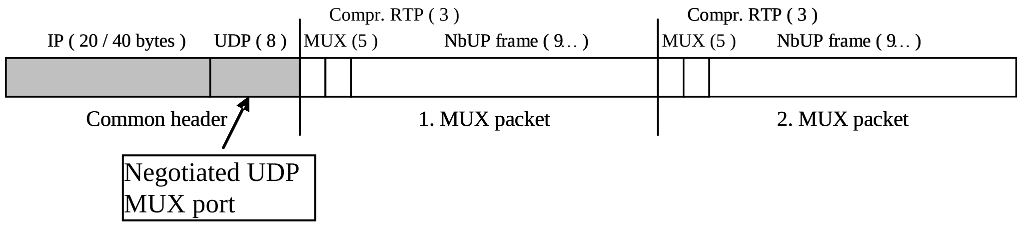

Figure 4: Example of multiplexed packet with two RTP frames and compressed RTP headers

5.1.2.2 RTP Header Reduction for VoIP transport with RTP framing

The principles specified in sub-clause 5.1.2.1 are re-used for VoIP transport with RTP framing. However in the more general case of non-NbFP payload types, any of the remaining other RTP header fields may also change, depending on the scenario, for instance:

- If RTP level multiplexing is used, for instance in a conference call, SSRC and CSRC and CC may change.

- If several codecs are negotiated, e.g. DTMF of DTX in addition to a speech codec, the payload type may change.

- The usage of the M bit depends upon the payload. The M bit is used for instance in VoIP context when speech resumes after silence, or for DTMF digits (RFC 2833 [5]).

- The usage of the X bit and RTP header extensions also depend upon the payload.

It has been proposed to send the full RTP header if any of the fields in the compressed header changes. This procedure bears the risk that sender and receiver get out of synch if packets are lost. It would therefore be beneficial to send several full RTP packets after a change and also transmit full RTP packets in regular intervals. These procedures are suggested to be the subject of a separate study.

These procedures are not suited for frequent changes. In such scenarios the usage of an uncompressed RTP header is recommended.

The M bit and the PT are considered to be of particular importance for point to point VoIP bearers, and therefore the format in figure 5 has been proposed. Procedures how to negotiate the possible usage of this format are not yet defined. This format is suggested to be the subject of a separate study.

Bits								Number of Octets	
7	6	5	4	3	2	1	0		
Source IP, Dest IP, …								20/40	IP
Source Port, Dest Port=<MUX UDP port>, Length, …								8	UDP
T=1	Mux ID = (Destination UDP Port of multiplexed PDU) / 2							2	Multiplex Header
Length Indicator (LI) = n+ 4								1	
R	Source ID = (Source UDP Port of multiplexed PDU) / 2							2	
Sequence Number (SN)								1	Compressed RTP header
Timestamp (TS)								2	
M	Payload Type (PT)							1	
RTP payload								n	RTP NbFP Payload
Multiplex Header								5	Multiplex Header
Compressed RTP header								4	Compressed RTP header
RTP payload								m	RTP NbFP Payload
…									

Figure 5: Compressed RTP header for VoIP transport

5.2 Evaluation of Multiplexing Efficiency

5.2.1 Efficiency of UDP Port Multiplexing

The bandwidth decreases in different cases are illustrated in table 1. In PoS cases the network is assumed to use double MPLS framing (VPN and traffic type differentiation) and Ethernet is assumed to use VLAN tag.

Table 1: Bandwidths with AMR12.2 (60 % activity factor) without and with multiplexing (2 or 10 RTP frames, common IP/UDP header)

	PoS, IPv4	PoS, IPv6	Eth, IPv4	Eth, IPv4
BW ref	22,88 kbps	28,08 kbps	29,90 kbps	35,10 kbps
BW, 2 pkts	18,59 kbps	21,19 kbps	22,10 kbps	24,70 kbps
Decrease	19 %	24 %	26 %	30 %
BW, 10 pkts	14,12 kbps	14,64 kbps	14,82 kbps	15,34 kbps
Decrease	38 %	48 %	50 %	56 %

5.2.2 Efficiency of RTP Header Compression

The bandwidth decreases with the same assumptions as in the normal case are shown in table 2.

Table 2: Bandwidths with AMR12.2 (60 % activity factor) without and with multiplexing (2 or 10 RTP frames, common IP/UDP header) with compressed RTP header

	PoS, IPv4	PoS, IPv6	Eth, IPv4	Eth, IPv4
BW ref	22,88 kbps	28,08 kbps	29,90 kbps	35,10 kbps
BW, 2 pkts	16,25 kbps	18,85 kbps	19,76 kbps	22,36 kbps
Decrease	29 %	33 %	34 %	36 %
BW, 10 pkts	11,78 kbps	12,30 kbps	12,48 kbps	13,00 kbps
Decrease	48 %	56 %	58 %	63 %

6 Signalling Extensions

6.1 Multiplex negotiation via IuFP User Plane Protocol

The bearer initialization phase in both Nb- and Iu-interfaces include mandatory messages for the support mode that is used e.g. for speech traffic. Nb/Iu UP PDU Type 14 is used at initialization and the messages include spare extension fields (both initialization and acknowledgement frames) that can be used for any additional function and this field is proposed to be used for multiplexing applicability detection. The field is proposed to be one byte long from which two first bits are used for multiplexing detection and the rest six are spared for future use (figure 4). The transparent mode in the Iu-interface would not support multiplexing since it has no initialization phase but it is not used for speech applications and these are the most common traffic type in CS domain.

<table>
<tr><td colspan="8">Bits</td><td rowspan="2">Number of Octets</td><td></td></tr>
<tr><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td>0</td><td></td></tr>
<tr><td colspan="4">PDU Type (=14)</td><td colspan="2">Ack/Nack (=0. I.e. Procedure)</td><td colspan="2">PDU Type 14 Frame Number</td><td>1</td><td rowspan="2">Frame Control Part</td></tr>
<tr><td colspan="4">Iu UP Mode version</td><td colspan="4">Procedure Indicator (=0)</td><td>1</td></tr>
<tr><td colspan="6">Header CRC</td><td colspan="2">Payload CRC</td><td rowspan="2">2</td><td rowspan="2">Frame Checksum part</td></tr>
<tr><td colspan="8">Payload CRC</td></tr>
<tr><td colspan="3">Spare</td><td>TI</td><td colspan="3">Number of subflows per RFCI (N)</td><td>Chain Ind</td><td>1</td><td rowspan="14">Frame payload part</td></tr>
<tr><td>LRI</td><td>LI</td><td colspan="6">1st RFCI</td><td>1</td></tr>
<tr><td colspan="8">Length of subflow 1</td><td>1 or 2 (dep. LI)</td></tr>
<tr><td colspan="8">Length of subflow 2 to N</td><td>(N-1)x(1 or 2)</td></tr>
<tr><td>LRI</td><td>LI</td><td colspan="6">2nd RFCI</td><td>1</td></tr>
<tr><td colspan="8">Length of subflow 1</td><td>1 or 2 (dep. LI)</td></tr>
<tr><td colspan="8">Length of subflow 2 to N</td><td>(N-1)x(1 or 2)</td></tr>
<tr><td colspan="8">…</td><td></td></tr>
<tr><td colspan="4">IPTI of 1st RFCI</td><td colspan="4">…</td><td rowspan="2">0 or M/2 (M: Number of RFCIs in frame). Ended by 4 padding bits if M is odd.</td></tr>
<tr><td colspan="4">…</td><td colspan="4">IPTI of Mth RFCI or Padding</td></tr>
<tr><td colspan="8">Iu UP Mode Versions supported (bitmap)</td><td>2</td></tr>
<tr><td colspan="4">Data PDU type</td><td colspan="4">Spare</td><td>1</td></tr>
<tr><td colspan="6">Spare</td><td>MUX & RTP c</td><td>MUX</td><td>1</td></tr>
</table>

Figure 4: UP PDU Type 14 used for initialization with one byte spare extension field for multiplexing detection (last octet)

When an MGW or RNC supports multiplexing it sets the first bit (MUX in figure 4) to 1 in the spare extension field of the initialization frame and from that bit the receiving node knows that multiplexing can be used. If the receiving node supports multiplexing it replies in the same way with the first bit set to 1 in the spare extension field of the positive acknowledgement message and again the other end knows that multiplexing can be used. If the receiving node does not support it just ignores the spare extension in the initialization and sends a regular acknowledgement. The MGW or RNC that started the initialization knows then not to use multiplexing.

Since a node may support multiplexing but not RTP header compression there must be separate initializations. While the first bit stands for normal multiplexing, the RTP header compression possibility is indicated with the second bit (MUX & RTP c in figure 4). The destination node can now reply in three ways. Responding with the second bit it indicates that the RTP header compression may be used. It may however reply also with the first bit meant for normal multiplexing or reply without any multiplexing indications.

For Nb interface there already is a standardized protocol for bearer control, IP Bearer Control Protocol (IPBCP), and it could be used also for detecting multiplexing applicability. IPBCP however cannot be used for Iu-interface and therefore UP initialization as a more common solution is better for initializing multiplexing. In general the applicability detection can be seen as a migration phase function, which could be left out when all nodes support multiplexing. After that multiplexed packets could be always detected based on the UDP port (2002 for normal multiplexed packets and 2004 for multiplexed packets with RTP header compression).

6.2 Multiplex negotiation using IPBCP

If bandwidth saving solution is used in other network which doesn't have user plane but it uses IPBCP to bear control, it is proposed to use IPBCP to multiplexing detection.

An MGW wishing to apply multiplexing shall add the parameter in the SDP offer / IPBCP Request. The initial MGW indicates its supporting format list. If the receiving MGW supports at least one of the formats and is willing to use multiplexing it shall return its selected format.

When IPBCP is used for multiplexing detection it is proposed to use "fmtp" and two new parameters should be defined:

"IPFmts": define list of formats proposed to be used in the call; 0: format 0; 1: format 1.

EXAMPLE: MGW A and MGW B support the new transport option.

```
IPBCP Request: (MGW A -> MGW B)
      m=audio 49160 RTP/AVP 97
      a=rtpmap:97 VND.3GPP.IUFP/16000
      a=fmtp:97 IPFmts={0,1}; UPC=T

IPBCP Response: (MGW B -> MGW A)
      m=audio 49300 RTP/AVP 97
      a=rtpmap:97 VND.3GPP.IUFP/16000
      a=fmtp:97 IPFmts={1} ; UPC=T
```

6.3 Multiplex negotiation using RTCP

According to 3GPP TS 29.414 [1] and 3GPP TS 25.414 [3], RTCP (RFC 3550 [4]) may be used at the Nb interface with BICC signalling and the Iu interface. The usage of RTCP is optional on the Nb interface and otherwise recommended according to RFC 3550 [4]. Thus, RTCP is the only bearer signalling protocol applicable in all scenarios of interest, including a SIP-I based 3GPP Cs domain, and the Iu interface. For host supporting multiplexing, the usage of RTCP would need to be mandated.

RTCP allows for the addition of application specific new packet types, which might be defined by 3GPP and need IANA registration.

The application specific RTCP packets may be added to compound RTCP packets transferred within RTCP messages.

The following contents of a new "Multiplexing" RTCP packet are proposed:

- Indication if multiplexing without RTP header compression is supported.
- Indication if multiplexing with RTP header compression is supported.
- Indication if multiplexing is selected.
- the local UDP Port number where to receive multiplexed data streams.

The following encoding is proposed for this RTCP packet:

<table>
<tr><td colspan="8">Bits</td><td rowspan="2">Number of Octets</td><td></td></tr>
<tr><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td>0</td><td></td></tr>
<tr><td colspan="2">V=2</td><td>P</td><td colspan="5">subtype</td><td>1</td><td rowspan="5">APP packet header</td></tr>
<tr><td colspan="8">PT=APP=204</td><td>1</td></tr>
<tr><td colspan="8">Length</td><td>2</td></tr>
<tr><td colspan="8">SSRC/CSRC</td><td>4</td></tr>
<tr><td colspan="8">Name(ASCII)</td><td>4</td></tr>
<tr><td>MUX</td><td>CP</td><td colspan="2">Selection</td><td colspan="4">Reserved=0000</td><td rowspan="2">2</td><td rowspan="3">Application dependent data</td></tr>
<tr><td colspan="8">Reserved=00000000</td></tr>
<tr><td>Reserved=0</td><td colspan="7">Local MUX UDP port / 2</td><td>2</td></tr>
</table>

Figure 5: RTCP Multiplexing packet

The APP packet header includes :

- version (V), 2 bits
  Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. RTP Version 2 shall be used.

- padding (P), 1 bit
  As specified in RFC 3550 [4].

- subtype, 5 bits.
  May be used as a subtype to allow a set of APP packets to be defined under one unique name, or for any application-dependent data. The following subtype shall be used :
  00001 : RTCP Multiplexing packet

- packet type (PT), 8 bits.
  Contains the constant 204 to identify this as an RTCP APP packet

- length, 16 bits.
  As specified in RFC 3550 [4]. The length of this RTCP packet in 32-bit words minus one, including the header and any padding. (The offset of one makes zero a valid length and avoids a possible infinite loop in scanning a compound RTCP packet, while counting 32-bit words avoids a validity check for a multiple of 4.)

- SSRC/CSRC, 32 bits.
  As specified in RFC 3550 [4].

- name, 32 bits.
  A name chosen by the person defining the set of APP packets to be unique with respect to other APP packets this application might receive. The application creator might choose to use the application name, and then coordinate the allocation of subtype values to others who want to define new packet types for the application. Alternatively, it is RECOMMENDED that others choose a name based on the entity they represent, then coordinate the use of the name within that entity. The name is interpreted as a sequence of four ASCII

characters, with uppercase and lowercase characters treated as distinct.
Set to '3GPP' in our case.

The application-dependent data includes :

- multiplexing bit (MUX), 1 bit
  Indicates whether multiplexing without RTP header compression is supported or not by the sender of the RTCP packet : set to 0 if not supported, set to 1 if supported.

- multiplexing with RTP header compression bit (CP), 1 bit
  Indicates whether multiplexing with RTP header compression is supported or not by the sender of the RTCP packet : set to 0 if not supported, set to 1 if supported.

- Selection bits, 2 bits
  Indicates whether the sender of the RTCP packet has selected to apply multiplexing with or without header compression for the user plane packets that it sends on this connection. The following values are defined:
  00: no multiplexing is applied
  01: multiplexing is applied without RTP header compression
  10: multiplexing is applied with RTP header compression
  11: reserved

- Local MUX UDP port, 15 bits :
  Local UDP port where the sender demands to receive multiplexed data streams without RTP header compression if the MUX bit indicates that multiplexing without RTP header compression is supported. The value shall be the same as the local MUX UDP port divided by two. This parameter shall be ignored by the receiver of the RTCP Multiplexing packet if the MUX and CP bits indicate that multiplexing is not supported.

- Reserved bits:
  Extension bits may be added in the RTCP Multiplexing packet in future releases. Reserved bits shall be set to 0 in sent RTCP Multiplexing packet of this relaese and shall be ignored in incoming RTCP Multiplexing packets

Extension fields may be added in the RTCP Multiplexing packet in future releases. They shall be ignored by a MGW implementing an earlier version of the specification.

The following procedure is proposed:

When setting up a new user plane connection, both peer MGWs start to send data without applying multiplexing and indicate their readiness to receive multiplexed data streams by including the new RTCP Multiplexing package in the initial and all subsequent RTCP packets they send. A MGW shall always announce the same multiplexing capabilities and the same UDP port where to receive multiplexed data streams in all the RTCP Multiplexing packets it sends for a given RTP session. MGWs should preferably send their initial RTCP packet at the very beginning of the RTP session to be able to apply multiplexing as soon as possible. A MGW sending a Multiplexing packet indicating support of multiplexing shall be ready to receive multiplexed packets on any of the announced UDP ports. A single UDP port for multiplexing shall be used per destination IP address.

A MGW receiving an RTCP packet, where the peer indicated its readiness to receive multiplexed data streams, may decide to apply multiplexing to send the corresponding RTP data streams towards the sender of the RTCP packet. If the MGW decides to apply multiplexing, it can immediately starts sending multiplexed data streams towards the corresponding UDP multiplexing port announced in the received RTCP Multiplexing packet. The MGW shall indicate in subsequent RTCP Multiplexing packets if it has decided to apply multiplexing with or without header compression for the given user connection.

NOTE: The information of the peer´s decision to apply multiplexing or not may be used by the receiving side e.g. to calculate traffic loads. This information may however be received quite much later than the actual time when the peer started applying multiplexing due to large RTCP period and unreliability of the RTCP signalling.

A MGW that does not receive RTCP or receives RTCP without the "Multiplexing" package shall continue to send data for the user connection without applying multiplexing.

A MGW that does not support multiplexing will ignore the unknown received RTCP "Multiplexing" package according to RTCP procedures and continue to send data for the user connection without applying multiplexing.

Sending of a RTCP Multiplexing packet indicating readiness to receive multiplexed data streams does not necessarily mean that the MGW is ready to send multiplexed data streams, i.e. multiplexing may be applied on a single or on both directions for a given RTP session.

The proposed procedures therefore enable smooth interoperations with hosts that do not sent RTCP or do no support multiplexing.

Some negative aspects of this proposal are that user connections are transported without multiplexing until the first RTCP package are received, the total multiplexing gain will be reduced by the time the user connection is applied without multiplexing. A two percent reduction of the multiplexing gain is expected.

NOTE: According to RFC 3550 [4], it is recommended to delay sending the first RTCP package by 2.5 seconds. Assuming a mean call duration of 2 minutes (120 seconds), the multiplexing gain will be reduced by 2 percent.

Also the proposal requires conditional support of RTCP in addition to the MUX feature for nodes that did not previously support RTCP; however RTCP is a protocol commonly and widely used together with RTP transport, recommended as per IETF RFC 3550, and that may also be used to compute quality statistics of the IP network.

6.4 Comparison of Advantages of Different Proposals for Multiplexing Negotiation.

Table 3 compares the applicability of the multiplexing negotiation proposal for use cases of interest. Only the proposal to use Multiplexing Negotiation using RTCP is applicable for all these use cases.

Table 3: Applicability of the multiplexing negotiation proposal for use cases of interest

	Nb Interface with BICC signalling at Nc interface	**Nb Interface with SIP-I signalling at Nc interface**	**Iu Interface**	**Other VoIP network**
Multiplex negotiation via IuFP	yes	no	yes	no
Multiplex negotiation using IPBCP	yes	no (see note)	no	no (see note)
Multiplex negotiation using RTCP	yes	yes	yes	yes
NOTE: Similar principles could be used to extend SDP, possibly making the proposal applicable also to this scenario.				

The Proposal to use Multiplexing Negotiation via IuFP User Plane Protocol has minimal protocol impacts only. The other proposals require moderate protocol extension to SDP or RTCP, respectively.

The standardisation of all three proposals could be performed by 3GPP (with IANA registration of SDP or RTCP extensions). For the proposal "Multiplex negotiation using RTCP", a standardisation within IETF could be an alternative to achieve a more widespread support.

The Proposal to use Multiplex negotiation using IPBCP has the advantage that it performed before the user plane connection is established, thus avoiding the need of a reconfiguration during the ongoing call and also enabling an improved resource management.

The Proposal "Multiplexing Negotiation using RTCP" reduces the multiplexing gain around 2 percent and requires a support of RTCP.

7 Summary of Conclusions

The present document defines a solution allowing to achieve significant bandwidth reductions in an IP network by multiplexing packets with the same destination address and DiffServ class, while taking care to preserve the optional capability to avoid RTPXtalk situations. Extra bandwidth reduction can further be obtained by supporting RTP header compression.

A format of compressed RTP headers has been defined for NbFP transport. Suggestions for the usage of compressed RTP headers in other VoIP transport with RTP framing are also included, but are suggested to be the subject of a separate study.

The transport formats specified in subclause 5.1.1 and 5.1.2.1 are agreed to be added as options for the Nb interface within the scope of 3GPP Rel-7 work. For being the only bearer signalling protocol applicable in all scenarios of interest, including a SIP-I based 3GPP Cs domain, and the Iu interface, and a protocol widely used in conjunction with RTP transport, RTCP is agreed to be retained for negotiation of multiplexing and RTP header compression. By recommending MGWs to send their initial RTCP packet at the very beginning of the RTP session, its use does not entail bandwidth gains reduction.

Annex A: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
05/2006	CT3#40				TR Template agreed		0.0.0
05/2006	CT3#40				Content according to agreed TDOC C3-060276	0.0.0	0.1.0
09/2006	CT3#41				The following agreed TDOCs have been included: C3-060425, C3-060507	0.1.0	0.2.0
11/2006	CT3#42				The following agreed TDOCs have been included: C3-060685, C3-060863	0.2.0	0.3.0
12/2006	TSG#33	CP-060639			Editorial update by MCC for presentation to TSG CT for information	0.3.0	1.0.0
02/2007	CT3#43				The following agreed TDOCs have been included: C3-070157, C3-070197	1.0.0	1.1.0
02/2007	TSG#35	CP-070104			Editorial update by MCC for presentation to TSG CT for approval	1.1.0	2.0.0
03/2007					MCC update to version 7.0.0 after approval at TSG CT#35	2.0.0	7.0.0
06/2007	TSG#36	CP-070424	001	1	Correction of editorial mistake in figure for RTP compression	7.0.0	7.1.0