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3RD GENERATION
PARTNERSHIP
PROJECT 2
"3GPP2"

10 *Header Stripping and Generation*

11 12 *Stage 1 Requirements*

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1 **1 INTRODUCTION**

2 This document specifies the requirements for and operations of the header
3 stripping and generation feature from the perspective of the user and the system
4 operator.

5 The objective is to define and to standardize the functionality of this feature that
6 can be incorporated into the operations of CDMA2000 based wireless
7 telecommunications networks.

8 The scope of this feature is in support of QoS-enabled packet data service. This
9 feature may also be used to support the ALLIP multimedia domain, although
10 ALLIP is not the only application. For example, header stripping and generation
11 may be used to transport a voice RTP stream that was established by any
12 signaling mechanism available to the mobile node on the packet data network.

2 REFERENCES

The document references that are applicable to this specification include the following:

2.1 NORMATIVE REFERENCES

- **C.S0001** Introduction to CDMA2000 Standards for Spread Spectrum Systems
- **C.S0002** Physical Layer Standard for CDMA2000 Spread Spectrum Systems
- **C.S0003** Medium Access Control (MAC) Standard for CDMA2000 Spread Spectrum Systems
- **C.S0004** Signaling Link Access Control (LAC) Standard for CDMA2000 Spread Spectrum Systems
- **C.S0005** Upper Layer (Layer 3) Signaling Standard for CDMA2000 Spread Spectrum Systems
- **A.S0001** Access Network Interfaces Technical Specification
- **P.S0001** Wireless IP Network Standard
- **[Shulzrinne96]** Schulzrinne, H., Casner, S., Frederick, R., and Jacobson, V., "RTP: A Transport Protocol for Real-Time Applications," RFC 1889, January 1996.
- **[TIA-IS127]** Telecommunications Industry Association, "Enhanced Variable Rate Codec, Speech Service 3 for Wideband Spread Spectrum Digital Systems," TIA/EIA/IS-127, February 1997.
- **[TIA-SMV]** Telecommunications Industry Association, "Selectable Mode Vocoder Service Option for Wideband Spread Spectrum Communication Systems," TIA PN4575, 3GPP2 C.P9001, 1997.

2.2 INFORMATIVE REFERENCES

- **[Bormann01]** Bormann, C. (ed.), "RObust Header Compression (ROHC)," RFC 3095, March 2001.
- **[ETSI-AMR]** European Telecommunications Standards Institute, "Adaptive Multi-Rate (AMR) Speech Transcoding," 3G TS 26.090, February 2000.
- **[McCann01]** McCann, P., Hiller, T. "Requirements and Architecture for Header Stripping and Generation", draft-mccann-rohc-gehcoarch-02, June 2001. Work In Progress.

1 3 DEFINITIONS AND ABBREVIATIONS

2 The terms and abbreviations that are used within this specification are defined as
3 follows:

4

A/D	Analog-to-Digital
BS	Base Station
CSRC	Contributing Source
D/A	Digital-to-Analog
ECN	Explicit Congestion Notification
EVRC	Enhanced Variable-Rate Codec
GPRS	General Packet Radio Service
HDLC	High-level Data Link Control
HSG	Header Stripping and Generation
IP	Internet Protocol
IPv4	Internet Protocol, Version 4
IPv6	Internet Protocol, Version 6
LLC	Logical Link Control
LSB	Least Significant Bits
MS	Mobile Station
MTU	Maximum Transmission Unit
PCF	Packet Control Function
PDCP	Packet Data Convergence Protocol
PDSN	Packet Data Serving Node
PPP	Point-to-Point Protocol
RC	Rarely Changing
ROHC	RObust Header Compression
RTP	Real-Time Protocol
RTCP	Real-Time Control Protocol
SMV	Selectable Mode Vocoder
SRTP	Secure RTP
SSRC	Sending Source

UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications Service
VOIP	Voice over IP

1

2 **4 GENERAL FEATURE DESCRIPTION**

3 Efficient transmission of voice over wireless links requires significant engineering
4 effort. Because of the high cost of bandwidth on such links, special techniques
5 for compression of voice data and its transmission over the air have been
6 developed. The compression techniques and the wireless physical layers have
7 been co-designed for maximum spectral efficiency and human perceptual
8 euphony.

9 Voice over IP (VOIP) applications should be able to leverage this engineering
10 effort when used over wireless links. This document describes Stage 1
11 requirements for a “header stripping and generation” approach to this problem in
12 order to enable the end-to-end service model while achieving maximum spectral
13 efficiency and simplicity of implementation.

14 **5 DETAILED FUNCTIONALITY REQUIREMENTS**

15 Voice over IP (VOIP) promises to change radically the way that telephony
16 services are built and delivered. Integration of voice with the Internet will not just
17 be a change in the way traffic is carried; rather, new types of services will be
18 made possible by the integration of voice with existing Internet applications such
19 as the World Wide Web and e-mail. The key to these new services will be a
20 platform that offers open programmability while offering a transport for VOIP in
21 an integrated, robust, and efficient way.

22 Wireless links offer great challenges to the transport of voice traffic, and
23 significant engineering effort has gone into making them efficient for circuit voice
24 applications. New voice compression algorithms (“codecs”), such as EVRC
25 [TIA-IS127], SMV [TIA-SMV], or AMR [ETSI-AMR] have been developed to
26 minimize the amount of data that must be carried, and special over-the-air
27 channels have been implemented to carry these codecs with a minimum of
28 overhead bits and minimal latency.

1 VOIP flows will be carried inside the Real-Time Protocol (RTP) [Shulzrinne96]
2 on wired links. However, for wireless links, the situation is less clear. The
3 limited bandwidth of wireless links makes it impossible to transmit the entire
4 IP/UDP/RTP header with every packet, as the overhead would be prohibitive. It
5 is possible to compress these headers by transmitting only updates to the fields
6 that change rather than the entire header [Bormann01], but these compression
7 schemes can never entirely eliminate the overhead due to RTP. Even when the
8 header is compressed down to one byte per frame on average, the impact on
9 spectral capacity is significant. Also, the variable-sized frames produced by these
10 compression protocols are unsuitable for typical wireless links that support only a
11 limited number of frame sizes.

12 The fundamental reason why these schemes cannot achieve the same efficiency as
13 circuit voice is that they operate asynchronously, discarding information that is
14 available at the physical channel layer, including the real-time nature of the
15 traffic, which can assist in reconstructing the RTP header. They must instead
16 explicitly encode this information with header bits. This document describes a
17 feature that allows such real-time information to be used in the reconstruction of
18 the RTP header.

19 Also, existing schemes assume that decompressed headers generated for each
20 packet must be identical to the headers that were compressed. If this can not be
21 achieved for a particular packet, existing schemes discard the packet. However,
22 many applications, especially those that are only concerned with transmission and
23 playback of voice, can tolerate some amount of skew in the reproduced RTP
24 headers. When a compressor/decompressor pair can make these assumptions,
25 very simple and efficient header compression can be performed. The architecture
26 should allow applications to indicate their ability to tolerate such skew. This
27 allows us to implement a form of header compression that makes use of existing
28 circuit voice implementations with minimal changes; we refer to this approach as
29 “header stripping and generation.”

30

31 **5.1 DETAILED FEATURE CHARACTERISTICS AND REQUIREMENTS**

32 This section gives requirements for the header stripping and generation service, in
33 the context of the necessary feature characteristics in terms of performance,
34 functionality, and simplicity. Requirements for header stripping and generation
35 are given within each discussion and labeled as **R_x**, where **x** is the requirement
36 number for tracking purposes.

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5.1.1 PERFORMANCE

Existing circuit voice implementations are co-engineered with the physical link to achieve maximum spectral efficiency. Here, "spectral efficiency" is the amount of bandwidth used by a voice call, given a particular level of voice quality. Voice quality is determined by many factors, including the encoding performed by a particular vocoder, the delay of the end-to-end path, the jitter experienced on that path, and the number of frames lost on the path. Header stripping and generation, while it will be used in a VOIP environment where some components of the end-to-end path may have unpredictable performance, should not itself contribute performance degradations when compared to existing circuit voice implementations. This can be expressed with the following requirements:

- R1** Low overhead. Header stripping and generation shall not increase the average transmitted bit rate when compared to circuit voice.
- R2** Low latency. Header stripping and generation shall not introduce any additional latency when compared to the transport of circuit voice.
- R3** Low jitter. Header stripping and generation shall not introduce any additional jitter when compared to the transport of circuit voice.

Note that R2 and R3 are requirements on the Header Stripping and Generation Service over-the-air only; it should not be interpreted to apply to the IP network that connects the access network to the remote VOIP peer. In particular, some amount of latency may be introduced as part of the mechanism to compensate for jitter in the Internet. Such jitter would not be present to the same extent in a circuit call that is connected through a circuit-switched network.

- R4** Low loss. Header stripping and generation shall not unnecessarily discard codec frames.

R4 requires the header stripping and generation service to deliver frames to the application whenever possible. When taken together, R1-R4 will assure that the header stripping and generation service has performance equivalent to that of circuit voice.

5.1.2 FUNCTIONAL CHARACTERISTICS

The basis for header stripping and generation is to use CDMA time as a proxy for the RTP sequence number. This is possible if the transmitter sends a vocoded frame for every epoch, as is done by the existing CDMA2000 vocoders even during silence intervals. Note that in 3G systems the mobile node transmits continuously even during silence so that the network may monitor power. Note also that these frames are not empty; they do carry information about the background noise components during silence, known as "comfort noise".

1 Header stripping and generation allows implementations that produce differences
2 between the reconstructed headers at the decompressor relative to the headers
3 seen by the compressor. It may introduce a small amount of “skew” between the
4 input and output frame streams. Note that for most VOIP applications, the RTP
5 sequence number and timestamp are primarily used to schedule frames for
6 playback over a relatively short interval. Implementations typically maintain a
7 playback buffer of a few frames, and place incoming voice samples into that
8 buffer based on their timestamp and sequence number. Based on a running
9 average of the buffer depth, frames are discarded or silence is inserted according
10 to whether the buffer is too full or is running low, respectively. Such a playback
11 buffer only needs the timestamps and sequence numbers to be relatively accurate;
12 that is, over short timescales, neighboring frames should have neighboring
13 timestamps and sequence numbers. Any small, fixed skew that is introduced into
14 the packet stream will be quickly corrected by the playback buffer mechanism.

15 However, not every application will be able to tolerate such skew. Some
16 applications will impose semantics on header fields that require them to be
17 transported bit-for-bit, or “transparently,” over the air. Therefore, the following
18 requirements are imposed:

- 19
- 20 **R5** The header stripping and generation service shall inform applications
21 that it is operating.
 - 22 **R6** Applications shall be able to turn off header stripping and generation so
23 that header bits are transported transparently over the air, although with
24 possibly reduced efficiency.
 - 25 **R7** Semantics preserving. Header stripping and generation shall only be
26 used when application semantics can tolerate the skew introduced to
27 the RTP header fields.

28 **5.1.3 SIMPLICITY**

29 A major goal of header stripping and generation is to support transport of voice
30 over existing cellular voice channels with little or no changes on the supporting
31 radio access equipment. Allowing a solution to completely strip out the header,
32 transmitting only voice data on this channel, will significantly aid that goal. By
33 not imposing any new format requirements on the vocoded frames, we allow
34 development of future codecs to proceed with maximum flexibility.

35 The simplicity of the supporting header compression state machine must also be
36 considered. Wireless devices are likely to be limited in both power and memory
37 budgets. Network access servers, while they will be implemented on larger
38 footprint equipment, will need to support large numbers of attached devices and
39 so scalability is a key issue. By decoupling the header initialization and updates
40 from the synchronous voice traffic channel, it may be possible to achieve
41 significant simplifications in the header compression protocol state machine.

42 The discussion above leads to the following requirements:

- 1 **R8** Simplicity. Header stripping and generation shall not require the BS to
2 implement context update logic.
- 3 **R9** Circuit re-use. Header stripping and generation shall use the existing
4 circuit voice transport mechanism with only the minimal changes
5 required to initialize and connect the service.
- 6 **R10** Codec transparency. Header stripping and generation shall not impose
7 any additional semantics on codec data bits.
- 8 **R11** Context update. Any signaling to update header generation context
9 shall take place out-of-band with respect to the voice codec data.

10 **5.2 CALL MODEL**

11 From the user's perspective, header stripping and generation is invoked by making
12 a voice-over-IP call using some voice application on the mobile terminal. The
13 mobile terminal uses the same IP address that it was assigned during packet data
14 registration to originate the call, and subsequent RTP packets carrying voice
15 codec information must appear to the rest of the network as if they originated
16 from this IP address. This document does not specify the choice of signaling
17 protocol or the placement of call feature servers in the network; header stripping
18 and generation is equally applicable to any such architecture.

19 Context for header stripping and generation, including full header information and
20 mapping of RTP streams onto service option connections, is signaled within PPP
21 between the mobile station and PDSN.

22 **5.3 APPLICABILITY TO TELECOMMUNICATIONS SERVICES**

23 Header stripping and generation is applicable to packet data telecommunications
24 services.

25 **5.4 TONES, ANNOUNCEMENTS, AND INDICATIONS TO THE USERS**

26 Not specified.

27 **5.5 NORMAL PROCEDURES WITH SUCCESSFUL OUTCOME**

- 28 ▪ Authorization
29 Header stripping and generation should be made generally available to any
30 user authorized for QoS-enabled packet data service.
- 31 ▪ De-Authorization
32 Header stripping and generation may be withdrawn at the subscriber's
33 request or for administrative reasons.
- 34 ▪ Registration
35 Header stripping and generation has no registration other than the
36 registration for QoS-enabled packet data service.

- 1 ▪ Deregistration
2 Header stripping and generation has no de-registration other than the de-
3 registration for QoS-enabled packet data service.
- 4 ▪ Activation
5 Header stripping and generation shall be activated upon configuration of
6 the MS for HSG operation. This configuration may be initiated by the user
7 or by an application running on the MS.
- 8 ▪ Deactivation
9 Header stripping and generation shall be de-activated when the MS is no
10 longer configured for HSG operation.
- 11 ▪ Invocation
12 Header stripping and generation may be invoked when an application on
13 the mobile terminal sets up a voice-over-IP call using a voice codec that is
14 matched to the air interface channel, and when the application is tolerant
15 of the skew that may be introduced by header stripping/generation
16 operation.
- 17 ▪ Normal Operation with Successful Outcome
18 While header stripping and generation is invoked, the RTP stream flows
19 end-to-end from the mobile terminal to the RTP endpoint in the network,
20 which may be a media gateway or correspondent node. The voice quality
21 perceived by the user must be identical to the quality achieved by a circuit
22 voice call using the same codec. Actions by the user to terminate the call
23 or change the routing of call legs are orthogonal to the header stripping
24 and generation feature and are outside the scope of this document.
- 25 ▪ Call Detail Record
26 During operation of header stripping and generation, normal accounting
27 rules defined in the Wireless IP Standard [P.S0001] apply. That document
28 specifies that duration of active traffic channel use and number of bytes
29 transferred be recorded. In particular to the header stripping and
30 generation service, the system should additionally record call detail
31 information for the following:
- 32 ▪ The fact that HSG operation was invoked
33 ▪ The duration of such invocation
34 ▪ The IP address of the RTP peer with which packets were exchanged
- 35 Also, “number of bytes transferred” shall be reported as bytes actually sent
36 and received over the air, that is, only the octets from the header-stripped
37 payloads. Call detail record information should be transmitted across the
38 R-P interface to the PDSN and propagated through the AAA infrastructure
39 to the home network.
- 40

1 **5.6 EXCEPTION PROCEDURES OR UNSUCCESSFUL OUTCOME**

2 ▪ Authorization
3 None identified.

4 ▪ De-Authorization
5 None identified.

6 ▪ Registration
7 None identified.

8 ▪ Deregistration
9 None identified.

10 ▪ Activation
11 None identified.

12 ▪ Deactivation
13 None identified.

14 ▪ Invocation
15 None identified.

16 ▪ Exceptions While Roaming
17 None identified.

18 ▪ Exceptions During Intersystem Handoff
19 None identified.

20 **5.7 ALTERNATE PROCEDURES**

21 None identified.

22 **5.8 INTERACTIONS WITH OTHER WIRELESS SERVICES**

23 ▪ Answer Hold (AH)
24 None identified.

25 ▪ Call Delivery (CD)
26 None identified.

27 ▪ Call Forwarding—Busy (CFB)
28 None identified.

29 ▪ Call Forwarding—Default (CFD)
30 None identified.

31 ▪ Call Forwarding—No Answer (CFNA)
32 None identified.

- 1 ▪ Call Forwarding—Unconditional (CFU)
2 None identified.
- 3 ▪ Call Transfer (CT)
4 None identified.
- 5 ▪ Call Waiting (CW)
6 None identified.
- 7 ▪ Calling Name Presentation (CNAP)
8 None identified.
- 9 ▪ Calling Name Restriction (CNAR)
10 None identified.
- 11 ▪ Calling Number Identification Presentation (CNIP)
12 None identified.
- 13 ▪ Calling Number Identification Restriction (CNIR)
14 None identified.
- 15 ▪ Conference Calling (CC)
16 None identified.
- 17 ▪ Do Not Disturb (DND)
18 None identified.
- 19 ▪ Emergency Services
20 None identified.
- 21 ▪ Flexible Alerting (FA)
22 None identified.
- 23 ▪ Message Waiting Notification (MWN)
24 None identified.
- 25 ▪ Mobile Access Hunting (MAH)
26 None identified.
- 27 ▪ Password Call Acceptance (PCA)
28 None identified.
- 29 ▪ Preferred Language (PL)
30 None identified.
- 31 ▪ Priority Access and Channel Assignment (PACA)
32 None identified.

- 1 ▪ Remote Feature Control (RFC)
2 None identified.
- 3 ▪ Selective Call Acceptance (SCA)
4 None identified.
- 5 ▪ Subscriber PIN Access (SPINA)
6 None identified.
- 7 ▪ Subscriber PIN Intercept (SPINI)
8 None identified.
- 9 ▪ Three-Way Calling (3WC)
10 None identified.
- 11 ▪ User Selective Call Forwarding (USCF)
12 None identified.
- 13 ▪ Voice Message Retrieval (VMR)
14 None identified.
- 15 ▪ Voice Privacy (VP)
16 None identified.

17 **6 HIGH LEVEL SYSTEM DESIGN**

18 **6.1 OVERVIEW**

19 Header stripping and generation will operate between a mobile station and PDSN.
20 The intervening equipment, including BTS, BS, and PCF, will transport the
21 header-stripped frames.

22 **6.2 AIR INTERFACE SUPPORT**

23 Header stripping and generation will require the following support from the air
24 interface:

- 25 ▪ Support for a new service option for the transport of header-stripped
26 frames.

27 This new service option should borrow heavily from the existing circuit voice
28 service option. That is, it will carry codec frames over a fundamental traffic
29 channel in a manner identical to the existing circuit voice. The important
30 difference is that the codec frames may be connected to a header
31 stripping/generation process instead of being routed directly to a vocoder.

32 **6.3 RAN SUPPORT**

33 Header stripping and generation will require the following support from the RAN:

- 1 ▪ Transport of header stripped frames to and from the frame selector in
- 2 the BS.
- 3 ▪ Frame selection and distribution performed in a manner identical to
- 4 that for circuit voice.
- 5 ▪ Delivery of the resulting codec frames to and from the PCF across the
- 6 A8 interface instead of the vocoder.

7 **6.4 INTERSYSTEM INTERFACE SUPPORT**

8 Header stripping and generation will require the following support from the
9 intersystem interfaces:

- 10 ▪ Establishment of an A8 and A10 connection corresponding to the new
- 11 service option.
- 12 ▪ Transport of header stripped frames through the PCF to and from the
- 13 PDSN.
- 14 ▪ Hard handoff of the service option from one RAN to another, which
- 15 may result in a change of PCF and/or PDSN.
- 16 ▪ Upon a change of PDSN, a PDSN-to-PDSN tunnel should be
- 17 maintained to transport the header stripped frames to and from the old
- 18 PDSN which contains the header stripping and generator process for
- 19 the call.

20 **6.5 ACTIONS ACROSS SYSTEM INTERFACES**

21 Actions described above should be carried out across the various system
22 interfaces. Also, the mobile terminal should signal the PDSN to indicate which
23 flows should be mapped onto the new service option connection and treated with
24 header stripping.
25