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J. Sjöberg
M. Westerlund
Ericsson
A. Lakanien
S. Wenger
Nokia
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RTP Payload Format for the Extended Adaptive Multi-Rate Wideband (AMR-WB+) Audio Codec

Status of This Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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Abstract

This document specifies a Real-time Transport Protocol (RTP) payload format for Extended Adaptive Multi-Rate Wideband (AMR-WB+) encoded audio signals. The AMR-WB+ codec is an audio extension of the AMR-WB speech codec. It encompasses the AMR-WB frame types and a number of new frame types designed to support high-quality music and speech. A media type registration for AMR-WB+ is included in this specification.

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[1.](#) Introduction

This document specifies the payload format for packetization of Extended Adaptive Multi-Rate Wideband (AMR-WB+) [[1](#)] encoded audio signals into the Real-time Transport Protocol (RTP) [[3](#)]. The payload format supports the transmission of mono or stereo audio, aggregating multiple frames per payload, and mechanisms enhancing the robustness of the packet stream against packet loss.

The AMR-WB+ codec is an extension of the Adaptive Multi-Rate Wideband (AMR-WB) speech codec. New features include extended audio bandwidth to enable high quality for non-speech signals (e.g., music), native support for stereophonic audio, and the option to operate on, and switch between, several internal sampling frequencies (ISFs). The primary usage scenario for AMR-WB+ is the transport over IP. Therefore, interworking with other transport networks, as discussed for AMR-WB in [[7](#)], is not a major concern and hence not addressed in this memo.

The expected key application for AMR-WB+ is streaming. To make the packetization process on a streaming server as efficient as possible, an octet-aligned payload format is desirable. Therefore, a bandwidth-efficient mode (as defined for AMR-WB in [[7](#)]) is not specified herein; the bandwidth savings of the bandwidth-efficient mode would be very small anyway, since all extension frame types are octet aligned.

The stereo encoding capability of AMR-WB+ renders the support for multi-channel transport at RTP payload format level, as specified for AMR-WB [[7](#)], obsolete. Therefore, this feature is not included in this memo.

This specification does not include a definition of a file format for AMR-WB+. Instead, it refers to the ISO-based 3GP file format [[14](#)], which supports AMR-WB+ and provides all functionality required. The 3GP format also supports storage of AMR, AMR-WB, and many other

multi-media formats, thereby allowing synchronized playback.

The rest of the document is organized as follows: Background information on the AMR-WB+ codec, and design principles, can be found in [Section 3](#). The payload format itself is specified in [Section 4](#). Sections [5](#) and [6](#) discuss congestion control and security considerations, respectively. In [Section 7](#), a media type registration is provided.

[2.](#) Definitions

[2.1.](#) Glossary

3GPP	- Third Generation Partnership Project
AMR	- Adaptive Multi-Rate (Codec)
AMR-WB	- Adaptive Multi-Rate Wideband (Codec)
AMR-WB+	- Extended Adaptive Multi-Rate Wideband (Codec)
CN	- Comfort Noise
DTX	- Discontinuous Transmission
FEC	- Forward Error Correction
FT	- Frame Type
ISF	- Internal Sampling Frequency
SCR	- Source-Controlled Rate Operation
SID	- Silence Indicator (the frames containing only CN parameters)
TFI	- Transport Frame Index
TS	- Timestamp
VAD	- Voice Activity Detection
UED	- Unequal Error Detection
UEP	- Unequal Error Protection

[2.2.](#) Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC 2119](#) [2].

[3.](#) Background of AMR-WB+ and Design Principles

The Extended Adaptive Multi-Rate Wideband (AMR-WB+) [[1](#)] audio codec is designed to compress speech and audio signals at low bit-rate and good quality. The codec is specified by the Third Generation Partnership Project (3GPP). The primary target applications are 1) the packet-switched streaming service (PSS) [[13](#)], 2) multimedia messaging service (MMS) [[18](#)], and 3) multimedia broadcast and multicast service (MBMS) [[19](#)]. However, due to its flexibility and robustness, AMR-WB+ is also well suited for streaming services in other highly varying transport environments, for example, the Internet.

[3.1](#). The AMR-WB+ Audio Codec

3GPP originally developed the AMR-WB+ audio codec for streaming and messaging services in Global System for Mobile communications (GSM) and third generation (3G) cellular systems. The codec is designed as an audio extension of the AMR-WB speech codec. The extension adds new functionality to the codec in order to provide high audio quality

for a wide range of signals including music. Stereophonic operation has also been added. A new, high-efficiency hybrid stereo coding algorithm enables stereo operation at bit-rates as low as 6.2 kbit/s.

The AMR-WB+ codec includes the nine frame types specified for AMR-WB, extended by new bit-rates ranging from 5.2 to 48 kbit/s. The AMR-WB frame types can employ only a 16000 Hz sampling frequency and operate only on monophonic signals. The newly introduced extension frame types, however, can operate at a number of internal sampling frequencies (ISFs), both in mono and stereo. Please see Table 24 in [[1](#)] for details. The output sampling frequency of the decoder is limited to 8, 16, 24, 32, or 48 kHz.

An overview of the AMR-WB+ encoding operations is provided as follows. The encoder receives the audio sampled at, for example, 48 kHz. The encoding process starts with pre-processing and resampling to the user-selected ISF. The encoding is performed on equally sized super-frames. Each super-frame corresponds to 2048 samples per channel, at the ISF. The codec carries out a number of encoding decisions for each super-frame, thereby choosing between different encoding algorithms and block lengths, so as to achieve a fidelity-optimized encoding adapted to the signal characteristics of the

source. The stereo encoding (if used) executes separately from the monophonic core encoding, thus enabling the selection of different combinations of core and stereo encoding rates. The resulting encoded audio is produced in four transport frames of equal length. Each transport frame corresponds to 512 samples at the ISF and is individually usable by the decoder, provided that its position in the super-frame structure is known.

The codec supports 13 different ISFs, ranging from 12.8 to 38.4 kHz, as described by Table 24 of [1]. The high number of ISFs allows a trade-off between the audio bandwidth and the target bit-rate. As encoding is performed on 2048 samples at the ISF, the duration of a super-frame and the effective bit-rate of the frame type in use varies.

The ISF of 25600 Hz has a super-frame duration of 80 ms. This is the 'nominal' value used to describe the encoding bit-rates henceforth. Assuming this normalization, the ISF selection results in bit-rate variations from 1/2 up to 3/2 of the nominal bit-rate.

The encoding for the extension modes is performed as one monophonic core encoding and one stereo encoding. The core encoding is executed by splitting the monophonic signal into a lower and a higher frequency band. The lower band is encoded employing either algebraic code excited linear prediction (ACELP) or transform coded excitation (TCX). This selection can be made once per transport frame, but must

obey certain limitations of legal combinations within the super-frame. The higher band is encoded using a low-rate parametric bandwidth extension approach.

The stereo signal is encoded employing a similar frequency band decomposition; however, here the signal is divided into three bands that are individually parameterized.

The total bit-rate produced by the extension is the result of the combination of the encoder's core rate, stereo rate, and ISF. The extension supports 8 different core encoding rates, producing bit-rates between 10.4 and 24.0 kbit/s; see Table 22 in [1]. There are 16 stereo encoding rates generating bit-rates between 2.0 and 8.0 kbit/s; see Table 23 in [1]. The frame type uniquely identifies the AMR-WB modes, 4 fixed extension rates (see below), 24 combinations of

core and stereo rates for stereo signals, and the 8 core rates for mono signals, as listed in Table 25 in [1]. This implies that the AMR-WB+ supports encoding rates between 10.4 and 32 kbit/s, assuming an ISF of 25600 Hz.

Different ISFs allow for additional freedom in the produced bit-rates and audio quality. The selection of an ISF changes the available audio bandwidth of the reconstructed signal, and also the total bit-rate. The bit-rate for a given combination of frame type and ISF is determined by multiplying the frame type's bit-rate with the used ISF's bit-rate factor; see Table 24 in [1].

The extension also has four frame types which have fixed ISFs. Please see frame types 10-13 in Table 21 in [1]. These four pre-defined frame types have a fixed input sampling frequency at the encoder, which can be set at either 16 or 24 kHz. Like the AMR-WB frame types, transport frames encoded utilizing these frame types represent exactly 20 ms of the audio signal. However, they are also part of 80 ms super-frames. Frame types 0-13 (AMR-WB and fixed extension rates), as listed in Table 21 in [1], do not require an explicit ISF indication. The other frame types, 14-47, require the ISF employed to be indicated.

The 32 different frame types of the extension, in combination with 13 ISFs, allows for a great flexibility in bit-rate and selection of desired audio quality. A number of combinations exist that produce the same codec bit-rate. For example, a 32 kbit/s audio stream can be produced by utilizing frame type 41 (i.e., 25.6 kbit/s) and the ISF of 32kHz ($5/4 * (19.2+6.4) = 32$ kbit/s), or frame type 47 and the ISF of 25.6 kHz ($1 * (24 + 8) = 32$ kbit/s). Which combination is more beneficial for the perceived audio quality depends on the content. In the above example, the first case provides a higher audio bandwidth, while the second one spends the same number of bits

on somewhat narrower audio bandwidth but provides higher fidelity. Encoders are free to select the combination they deem most beneficial.

Since a transport frame always corresponds to 512 samples at the used ISF, its duration is limited to the range 13.33 to 40 ms; see Table 1. An RTP Timestamp clock rate of 72000 Hz, as mandated by this specification, results in AMR-WB+ transport frame lengths of 960 to

2880 timestamp ticks, depending solely on the selected ISF.

Index	ISF	Duration(ms)	Duration(TS Ticks @ 72 kHz)
0	N/A	20	1440
1	12800	40	2880
2	14400	35.55	2560
3	16000	32	2304
4	17067	30	2160
5	19200	26.67	1920
6	21333	24	1728
7	24000	21.33	1536
8	25600	20	1440
9	28800	17.78	1280
10	32000	16	1152
11	34133	15	1080
12	36000	14.22	1024
13	38400	13.33	960

Table 1: Normative number of RTP Timestamp Ticks for each Transport Frame depending on ISF (ISF and Duration in ms are rounded)

The encoder is free to change both the ISF and the encoding frame type (both mono and stereo) during a session. For the extension frame types with index 10-13 and 16-47, the ISF and frame type changes are constrained to occur at super-frame boundaries. This implies that, for the frame types mentioned, the ISF is constant throughout a super-frame. This limitation does not apply for frame types with index 0-9, 14, and 15; i.e., the original AMR-WB frame types.

A number of features of the AMR-WB+ codec require special consideration from a transport point of view, and solutions that could perhaps be viewed as unorthodox. First, there are constraints on the RTP timestamping, due to the relationship of the frame duration and the ISFs. Second, each frame of encoded audio must maintain information about its frame type, ISF, and position in the super-frame.

The multi-rate encoding capability of AMR-WB+ is designed to preserve high audio quality under a wide range of bandwidth requirements and transmission conditions.

AMR-WB+ enables seamless switching between frame types that use the same number of audio channels and the same ISF. Every AMR-WB+ codec implementation is required to support all frame types defined by the codec and must be able to handle switching between any two frame types. Switching between frame types employing a different number of audio channels or a different ISF must also be supported, but it may not be completely seamless. Therefore, it is recommended to perform such switching infrequently and, if possible, during periods of silence.

[3.3.](#) Voice Activity Detection and Discontinuous Transmission

AMR-WB+ supports the same algorithms as AMR-WB for voice activity detection (VAD) and generation of comfort noise (CN) parameters during silence periods. However, these functionalities can only be used in conjunction with the AMR-WB frame types (FT=0-8). This option allows reducing the number of transmitted bits and packets during silence periods to a minimum. The operation of sending CN parameters at regular intervals during silence periods is usually called discontinuous transmission (DTX) or source controlled rate (SCR) operation. The AMR-WB+ frames containing CN parameters are called Silence Indicator (SID) frames. More details about the VAD and DTX functionality are provided in [\[4\]](#) and [\[5\]](#).

[3.4.](#) Support for Multi-Channel Session

Some of the AMR-WB+ frame types support the encoding of stereophonic audio. Because of this native support for a two-channel stereophonic signal, it does not seem necessary to support multi-channel transport with separate codec instances, as specified in the AMR-WB RTP payload [\[7\]](#). The codec has the capability of stereo to mono downmixing as part of the decoding process. Thus, a receiver that is only capable of playout of monophonic audio must still be able to decode and play signals originally encoded and transmitted as stereo. However, to avoid spending bits on a stereo encoding that is not going to be utilized, a mechanism is defined in this specification to signal mono-only audio.

Here, each frame is retransmitted once in the following RTP payload packet. $F(n-2) \dots f(n+4)$ denote a sequence of audio frames, and $p(n-1) \dots p(n+4)$ a sequence of payload packets.

The mechanism described does not require signaling at the session setup. In other words, the audio sender can choose to use this scheme without consulting the receiver. For a certain timestamp, the receiver may receive multiple copies of a frame containing encoded audio data or frames indicated as NO_DATA. The cost of this scheme is bandwidth and the receiver delay necessary to allow the redundant copy to arrive.

This redundancy scheme provides a functionality similar to the one described in [RFC 2198](#), but it works only if both original frames and redundant representations are AMR-WB+ frames. When the use of other media coding schemes is desirable, one has to resort to [RFC 2198](#).

The sender is responsible for selecting an appropriate amount of redundancy based on feedback about the channel conditions, e.g., in the RTP Control Protocol (RTCP) [\[3\]](#) receiver reports. The sender is also responsible for avoiding congestion, which may be exacerbated by redundancy (see [Section 5](#) for more details).

[3.6.2](#). Use of Frame Interleaving

To decrease protocol overhead, the payload design allows several audio transport frames to be encapsulated into a single RTP packet. One of the drawbacks of such an approach is that in case of packet loss several consecutive frames are lost. Consecutive frame loss normally renders error concealment less efficient and usually causes clearly audible and annoying distortions in the reconstructed audio. Interleaving of transport frames can improve the audio quality in such cases by distributing the consecutive losses into a number of isolated frame losses, which are easier to conceal. However, interleaving and bundling several frames per payload also increases end-to-end delay and sets higher buffering requirements. Therefore, interleaving is not appropriate for all use cases or devices. Streaming applications should most likely be able to exploit interleaving to improve audio quality in lossy transmission

conditions.

Note that this payload design supports the use of frame interleaving as an option. The usage of this feature needs to be negotiated in the session setup.

The interleaving supported by this format is rather flexible. For example, a continuous pattern can be defined, as depicted in Figure 2.

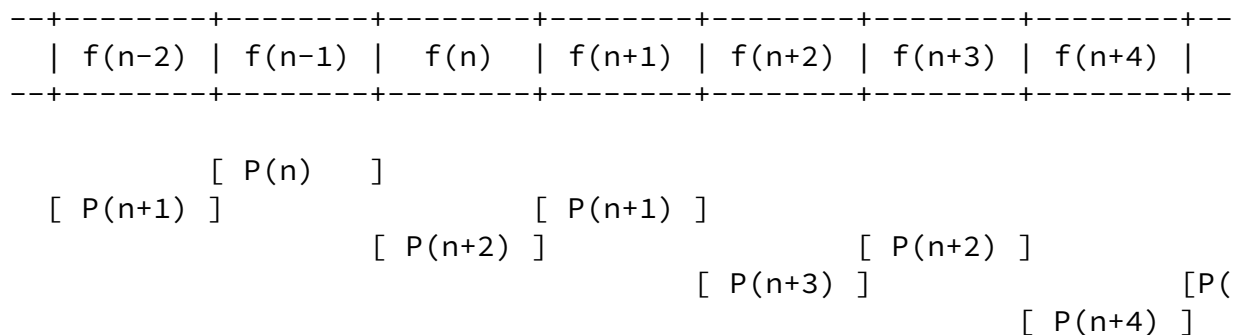


Figure 2: An example of interleaving pattern that has constant delay

In Figure 2 the consecutive frames, denoted $f(n-2)$ to $f(n+4)$, are aggregated into packets $P(n)$ to $P(n+4)$, each packet carrying two frames. This approach provides an interleaving pattern that allows for constant delay in both the interleaving and deinterleaving processes. The deinterleaving buffer needs to have room for at least three frames, including the one that is ready to be consumed. The storage space for three frames is needed, for example, when $f(n)$ is the next frame to be decoded: since frame $f(n)$ was received in packet $P(n+2)$, which also carried frame $f(n+3)$, both these frames are stored in the buffer. Furthermore, frame $f(n+1)$ received in the previous packet, $P(n+1)$, is also in the deinterleaving buffer. Note also that in this example the buffer occupancy varies: when frame $f(n+1)$ is the next one to be decoded, there are only two frames, $f(n+1)$ and $f(n+3)$, in the buffer.

[3.7.](#) AMR-WB+ Audio over IP Scenarios

Since the primary target application for the AMR-WB+ codec is streaming over packet networks, the most relevant usage scenario for this payload format is IP end-to-end between a server and a terminal,

as shown in Figure 3.

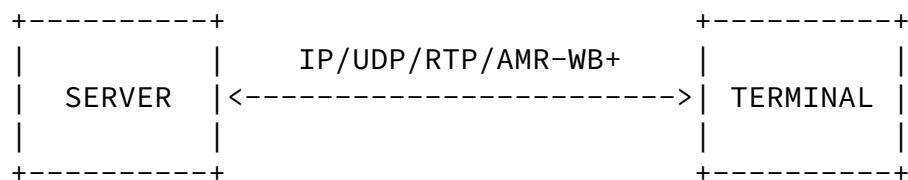


Figure 3: Server to terminal IP scenario

[3.8.](#) Out-of-Band Signaling

Some of the options of this payload format remain constant throughout a session. Therefore, they can be controlled/negotiated at the session setup. Throughout this specification, these options and variables are denoted as "parameters to be established through out-

of-band means". In [Section 7](#), all the parameters are formally specified in the form of media type registration for the AMR-WB+ encoding. The method used to signal these parameters at session setup or to arrange prior agreement of the participants is beyond the scope of this document; however, [Section 7.2](#) provides a mapping of the parameters into the Session Description Protocol (SDP) [6] for those applications that use SDP.

[4.](#) RTP Payload Format for AMR-WB+

The main emphasis in the payload design for AMR-WB+ has been to minimize the overhead in typical use cases, while providing full flexibility with a slightly higher overhead. In order to keep the specification reasonably simple, we refrained from defining frame-specific parameters for each frame type. Instead, a few common parameters were specified that cover all types of frames.

The payload format has two modes: basic mode and interleaved mode. The main structural difference between the two modes is the extension of the table of content entries with frame displacement fields when operating in the interleaved mode. The basic mode supports aggregation of multiple consecutive frames in a payload. The interleaved mode supports aggregation of multiple frames that are non-consecutive in time. In both modes it is possible to have frames encoded with different frame types in the same payload. The ISF must

remain constant throughout the payload of a single packet.

The payload format is designed around the property of AMR-WB+ frames that the frames are consecutive in time and share the same frame duration (in the absence of an ISF change). This enables the receiver to derive the timestamp for an individual frame within a payload. In basic mode, the deriving process is based on the order of frames. In interleaved mode, it is based on the compact displacement fields. The frame timestamps are used to regenerate the correct order of frames after reception, identify duplicates, and detect lost frames that require concealment.

The interleaving scheme of this payload format is significantly more flexible than the one specified in [RFC 3267](#). The AMR and AMR-WB payload format is only capable of using periodic patterns with frames taken from an interleaving group at fixed intervals. The interleaving scheme of this specification, in contrast, allows for any interleaving pattern, as long as the distance in decoding order between any two adjacent frames is not more than 256 frames. Note that even at the highest ISF this allows an interleaving depth of up to 3.41 seconds.

To allow for error resiliency through redundant transmission, the periods covered by multiple packets MAY overlap in time. A receiver MUST be prepared to receive any audio frame multiple times. All redundantly sent frames MUST use the same frame type and ISF, and MUST have the same RTP timestamp, or MUST be a NO_DATA frame (FT=15).

The payload consists of octet-aligned elements (header, ToC, and audio frames). Only the audio frames for AMR-WB frame types (0-9) require padding for octet alignment. If additional padding is desired, then the P bit in the RTP header MAY be set, and padding MAY be appended as specified in [\[3\]](#).

[4.1](#). RTP Header Usage

The format of the RTP header is specified in [\[3\]](#). This payload format uses the fields of the header in a manner consistent with that specification.

The RTP timestamp corresponds to the sampling instant of the first sample encoded for the first frame in the packet. The timestamp clock frequency SHALL be 72000 Hz. This frequency allows the frame duration to be integer RTP timestamp ticks for the ISFs specified in Table 1. It also provides reasonable conversion factors to the input/output audio sampling frequencies supported by the codec. See [Section 4.3.2.3](#) for guidance on how to derive the RTP timestamp for any audio frame beyond the first one.

The RTP header marker bit (M) SHALL be set to 1 whenever the first frame carried in the packet is the first frame in a talkspurt (see the definition of talkspurt in Section 4.1 of [9]). For all other packets, the marker bit SHALL be set to zero (M=0).

The assignment of an RTP payload type for the format defined in this memo is outside the scope of this document. The RTP profile in use either assigns a static payload type or mandates binding the payload type dynamically.

The media type parameter "channels" is used to indicate the maximum number of channels allowed for a given payload type. A payload type where channels=1 (mono) SHALL only carry mono content. A payload type for which channels=2 has been declared MAY carry both mono and stereo content. Note that this definition is different from the one in [RFC 3551](#) [9]. As mentioned before, the AMR-WB+ codec handles the support of stereo content and the (eventual) downmixing of stereo to mono internally. This makes it unnecessary to negotiate for the number of channels for reasons other than bit-rate efficiency.

[4.2.](#) Payload Structure

The payload consists of a payload header, a table of contents, and the audio data representing one or more audio frames. The following diagram shows the general payload format layout:

```
+-----+-----+-----+
| payload header | table of contents | audio data ...
+-----+-----+-----+
```

Payloads containing more than one audio frame are called compound

payloads.

The following sections describe the variations taken by the payload format depending on the mode in use: basic mode or interleaved mode.

[4.3.](#) Payload Definitions

[4.3.1.](#) Payload Header

The payload header carries data that is common for all frames in the payload. The structure of the payload header is described below.

```
 0 1 2 3 4 5 6 7
+---+---+---+---+
|   ISF   |TFI|L|
+---+---+---+---+
```

ISF (5 bits): Indicates the Internal Sampling Frequency employed for all frames in this payload. The index value corresponds to internal sampling frequency as specified in Table 24 in [1]. This field SHALL be set to 0 for payloads containing frames with Frame Type values 0-13.

TFI (2 bits): Transport Frame Index, from 0 (first) to 3 (last), indicating the position of the first transport frame of this payload in the AMR-WB+ super-frame structure. For payloads with frames of only Frame Type values 0-9, this field SHALL be set to 0 by the sender. The TFI value for a frame of type 0-9 SHALL be ignored by the receiver. Note that the frame type is coded in the table of contents (as discussed later); hence, the mentioned dependencies of the frame type can be applied easily by interpreting only values carried in the payload header. It is not necessary to interpret the audio bit stream itself.

L (1 bit): Long displacement field flag for payloads in interleaved mode. If set to 0, four-bit displacement fields are used to indicate interleaving offset; if set to 1, displacement fields of eight bits are used (see [Section 4.3.2.2](#)). For payloads in the

basic mode, this bit SHALL be set to 0 and SHALL be ignored by the receiver.

Note that frames employing different ISF values require encapsulation in separate packets. Thus, special considerations apply when generating interleaved packets and an ISF change is executed. In particular, frames that, according to the previously used interleaving pattern, would be aggregated into a single packet have to be separated into different packets, so that the aforementioned condition (all frames in a packet share the ISF) remains true. A naive implementation that splits the frames with different ISF into different packets can result in up to twice the number of RTP packets, when compared to an optimal interleaved solution. Alteration of the interleaving before and after the ISF change may reduce the need for extra RTP packets.

[4.3.2.](#) The Payload Table of Contents

The table of contents (ToC) consists of a list of entries, each entry corresponds to a group of audio frames carried in the payload, as depicted below.

```
+-----+-----+ ... +-----+
| ToC entry #1 | ToC entry #2 |           ToC entry #N |
+-----+-----+ ... +-----+
```

When multiple groups of frames are present in a payload, the ToC entries SHALL be placed in the packet in order of increasing RTP timestamp value (modulo 2^{32}) of the first transport frame the TOC entry represents.

[4.3.2.1.](#) ToC Entry in the Basic Mode

A ToC entry of a payload in the basic mode has the following format:

```

0                               1
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|F| Frame Type |   #frames   |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
```

F (1 bit): If set to 1, indicates that this ToC entry is followed by another ToC entry; if set to 0, indicates that this ToC entry is the last one in the ToC.

Frame Type (FT) (7 bits): Indicates the audio codec frame type used for the group of frames referenced by this ToC entry. FT designates the combination of AMR-WB+ core and stereo rate, one of the special AMR-WB+ frame types, the AMR-WB rate, or comfort noise, as specified by Table 25 in [1].

#frames (8 bits): Indicates the number of frames in the group referenced by this ToC entry. ToC entries with this field equal to 0 (which would indicate zero frames) SHALL NOT be used, and received packets with such a TOC entry SHALL be discarded.

[4.3.2.2](#). ToC Entry in the Interleaved Mode

Two different ToC entry formats are defined in interleaved mode. They differ in the length of the displacement field, 4 bits or 8 bits. The L-bit in the payload header differentiates between the two modes.

If L=0, a ToC entry has the following format:

0																1																2																3																																																															
0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9	0	1	2	3	4	5	6	7	8	9																																																																								
F																Frame Type																#frames																DIS1																...																DISi																...															
...																...																DISn																Padd																																																															

F (1 bit): See definition in 4.3.2.1.

Frame Type (FT) (7 bits): See definition in 4.3.2.1.

#frames (8 bits): See definition in 4.3.2.1.

DIS1...DISn (4 bits): A list of n (n=#frames) displacement fields indicating the displacement of the i:th (i=1..n) audio frame relative to the preceding audio frame in the payload, in units of frames. The four-bit unsigned integer displacement values may be between 0 and 15, indicating the number of audio frames in decoding order between the (i-1):th and the i:th frame in the payload. Note that for the first ToC entry of the payload, the value of DIS1 is meaningless. It SHALL be set to zero by a sender and SHALL be ignored by a receiver. This frame's location in the decoding order is uniquely defined by the RTP timestamp and TFI in the payload header. Note also that for subsequent ToC entries, DIS1 indicates the number of frames between the last frame of the previous group and the first frame of this group.

If L=1, a ToC entry has the following format:

The RTP Timestamp value for a frame SHALL be the timestamp value of

the first audio sample encoded in the frame. The timestamp value for a frame is derived differently depending on the payload mode, basic or interleaved. In both cases, the first frame in a compound packet has an RTP timestamp equal to the one received in the RTP header. In the basic mode, the RTP time for any subsequent frame is derived in two steps. First, the sum of the frame durations (see Table 1) of all the preceding frames in the payload is calculated. Then, this sum is added to the RTP header timestamp value. For example, let's

assume that the RTP Header timestamp value is 12345, the payload carries four frames, and the frame duration is 16 ms (ISF = 32 kHz) corresponding to 1152 timestamp ticks. Then the RTP timestamp of the fourth frame in the payload is $12345 + 3 * 1152 = 15801$.

In interleaved mode, the RTP timestamp for each frame in the payload is derived from the RTP header timestamp and the sum of the time offsets of all preceding frames in this payload. The frame timestamps are computed based on displacement fields and the frame duration derived from the ISF value. Note that the displacement in time between frame $i-1$ and frame i is $(DIS_i + 1) * \text{frame duration}$ because the duration of the $(i-1)$:th must also be taken into account. The timestamp of the first frame of the first group of frames (TS(1)) (i.e., the first frame of the payload) is the RTP header timestamp. For subsequent frames in the group, the timestamp is computed by

$$TS(i) = TS(i-1) + (DIS_i + 1) * \text{frame duration}, \quad 2 < i < n$$

For subsequent groups of frames, the timestamp of the first frame is computed by

$$TS(1) = TS_{\text{prev}} + (DIS_1 + 1) * \text{frame duration},$$

where TS_{prev} denotes the timestamp of the last frame in the previous group. The timestamps of the subsequent frames in the group are computed in the same way as for the first group.

The following example derives the RTP timestamps for the frames in an interleaved mode payload having the following header and ToC information:

RTP header timestamp: 12345
ISF = 32 kHz

Frame 1 displacement field: DIS1 = 0
Frame 2 displacement field: DIS2 = 6
Frame 3 displacement field: DIS3 = 4
Frame 4 displacement field: DIS4 = 7

Assuming an ISF of 32 kHz, which implies a frame duration of 16 ms, one frame lasts 1152 ticks. The timestamp of the first frame in the payload is the RTP timestamp, i.e., $TS(1) = \text{RTP TS}$. Note that the displacement field value for this frame must be ignored. For the second frame in the payload, the timestamp can be calculated as $TS(2) = TS(1) + (DIS2 + 1) * 1152 = 20409$. For the third frame, the timestamp is $TS(3) = TS(2) + (DIS3 + 1) * 1152 = 26169$. Finally, for the fourth frame of the payload, we have $TS(4) = TS(3) + (DIS4 + 1) * 1152 = 35385$.

[4.3.2.4](#). Frame Type Considerations

The value of Frame Type (FT) is defined in Table 25 in [1]. FT=14 (AUDIO_LOST) is used to denote frames that are lost. A NO_DATA (FT=15) frame could result from two situations: First, that no data has been produced by the audio encoder; and second, that no data is transmitted in the current payload. An example for the latter would be that the frame in question has been or will be sent in an earlier or later packet. The duration for these non-included frames is dependent on the internal sampling frequency indicated by the ISF field.

For frame types with index 0-13, the ISF field SHALL be set 0. The frame duration for these frame types is fixed to 20 ms in time, i.e., 1440 ticks in 72 kHz. For payloads containing only frames of type 0-9, the TFI field SHALL be set to 0 and SHALL be ignored by the receiver. In a payload combining frames of type 0-9 and 10-13, the TFI values need to be set to match the transport frames of type 10-13. Thus, frames of type 0-9 will also have a derived TFI, which is ignored.

[4.3.2.5](#). Other TOC Considerations

If a ToC entry with an undefined FT value is received, the whole packet SHALL be discarded. This is to avoid the loss of data synchronization in the depacketization process, which can result in a

severe degradation in audio quality.

Packets containing only NO_DATA frames SHOULD NOT be transmitted. Also, NO_DATA frames at the end of a frame sequence to be carried in a payload SHOULD NOT be included in the transmitted packet. The AMR-WB+ SCR/DTX is identical with AMR-WB SCR/DTX described in [5] and can only be used in combination with the AMR-WB frame types (0-8).

When multiple groups of frames are present, their ToC entries SHALL be placed in the ToC in order of increasing RTP timestamp value (modulo 2^{32}) of the first transport frame the TOC entry represents, independent of the payload mode. In basic mode, the frames SHALL be consecutive in time, while in interleaved mode the frames MAY not only be non-consecutive in time but MAY even have varying inter-frame distances.

[4.3.2.6](#). ToC Examples

The following example illustrates a ToC for three audio frames in basic mode. Note that in this case all audio frames are encoded using the same frame type, i.e., there is only one ToC entry.

```

      0                               1
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5
+---+---+---+---+---+---+---+---+
|0| Frame Type1 | #frames = 3 |
+---+---+---+---+---+---+---+---+
```

The next example depicts a ToC of three entries in basic mode. Note that in this case the payload also carries three frames, but three ToC entries are needed because the frames of the payload are encoded using different frame types.

```

      0                               1                               2                               3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|1| Frame Type1 | #frames = 1 |1| Frame Type2 | #frames = 1 |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|0| Frame Type3 | #frames = 1 |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
```

The following example illustrates a ToC with two entries in interleaved mode using four-bit displacement fields. The payload includes two groups of frames, the first one including a single frame, and the other one consisting of two frames.

```

0                               1                               2                               3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|1| Frame Type1 | #frames = 1 | DIS1 | padd |0| Frame Type2 |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| #frames = 2 | DIS1 | DIS2 |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```

4.3.3. Audio Data

Audio data of a payload consists of zero or more audio frames, as described in the ToC of the payload.

ToC entries with FT=14 or 15 represent frame types with a length of 0. Hence, no data SHALL be placed in the audio data section to represent frames of this type.

As already discussed, each audio frame of an extension frame type represents an AMR-WB+ transport frame corresponding to the encoding of 512 samples of audio, sampled with the internal sampling frequency specified by the ISF indicator. As an exception, frame types with index 10-13 are only capable of using a single internal sampling frequency (25600 Hz). The encoding rates (combination of core bit-rate and stereo bit-rate) are indicated in the frame type field of

the corresponding ToC entry. The octet length of the audio frame is implicitly defined by the frame type field and is given in Tables 21 and 25 of [1]. The order and numbering notation of the bits are as specified in [1]. For the AMR-WB+ extension frame types and comfort noise frames, the bits are in the order produced by the encoder. The last octet of each audio frame MUST be padded with zeroes at the end if not all bits in the octet are used. In other words, each audio frame MUST be octet-aligned.

4.3.4. Methods for Forming the Payload

The payload begins with the payload header, followed by the table of

contents, which consists of a list of ToC entries.

The audio data follows the table of contents. All the octets comprising an audio frame SHALL be appended to the payload as a unit. The audio frames are packetized in timestamp order within each group of frames (per ToC entry). The groups of frames are packetized in the same order as their corresponding ToC entries. Note that there are no data octets in a group having a ToC entry with FT=14 or FT=15.

[4.3.5.](#) Payload Examples

[4.3.5.1.](#) Example 1: Basic Mode Payload Carrying Multiple Frames Encoded Using the Same Frame Type

Figure 4 depicts a payload that carries three AMR-WB+ frames encoded using 14 kbit/s frame type (FT=26) with a frame length of 280 bits (35 bytes). The internal sampling frequency in this example is 25.6 kHz (ISF = 8). The TFI for the first frame is 2, indicating that the first transport frame in this payload is the third in a super-frame. Since this payload is in the basic mode, the subsequent frames of the payload are consecutive frames in decoding order, i.e., the fourth transport frame of the current super-frame and the first transport frame of the next super-frame. Note that because the frames are all encoded using the same frame type, only one ToC entry is required.

```

0                               1                               2                               3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ISF = 8 | 2 | 0 | 0 |   FT = 26   | #frames = 3 | f1(0...7) |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
```

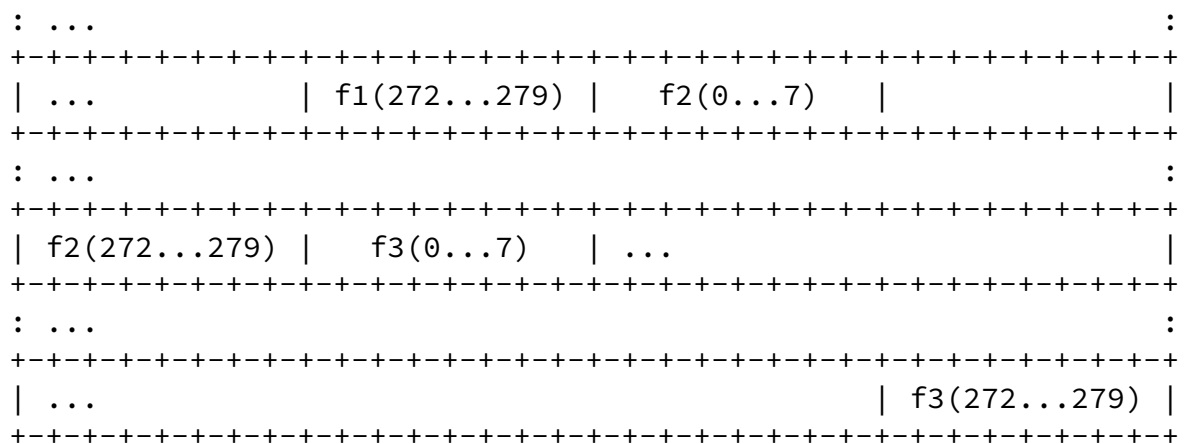



Figure 4: An example of a basic mode payload carrying three frames of the same frame type

[4.3.5.2](#). Example 2: Basic Mode Payload Carrying Multiple Frames Encoded Using Different Frame Types

Figure 5 depicts a payload that carries three AMR-WB+ frames; the first frame is encoded using 18.4 kbit/s frame type (FT=33) with a frame length of 368 bits (46 bytes), and the two subsequent frames are encoded using 20 kbit/s frame type (FT=35) having frame length of 400 bits (50 bytes). The internal sampling frequency in this example is 32 kHz (ISF = 10), implying the overall bit-rates of 23 kbit/s for the first frame of the payload, and 25 kbit/s for the subsequent frames. The TFI for the first frame is 3, indicating that the first transport frame in this payload is the fourth in a super-frame. Since this is a payload in the basic mode, the subsequent frames of the payload are consecutive frames in decoding order, i.e., the first and second transport frames of the current super-frame. Note that since the payload carries two different frame types, there are two ToC entries.


```

      0              1              2              3
    0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ISF=13 | 0 | 1|0|  FT = 47      | #frames = 4 |  DIS1 = 0      |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|  DIS2 = 18   |  DIS3 = 15   |  DIS4 = 10   |  f1(0...7)   |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
: ...                                                :
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ...                                                | f1(632...639) |  f2(0...7)   |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
: ...                                                :
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ...                                                | f2(632...639) |  f3(0...7)   |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
: ...                                                :
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ...                                                | f3(632...639) |  f4(0...7)   |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
: ...                                                :
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
| ...                                                | f4(632...639) |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```

Figure 6: An example of an interleaved mode payload carrying four frames at the same frame type

[4.4.](#) Interleaving Considerations

The use of interleaving requires further considerations. As presented in the example in [Section 3.6.2](#), a given interleaving pattern requires a certain amount of the deinterleaving buffer. This buffer space, expressed in a number of transport frame slots, is indicated by the "interleaving" media type parameter. The number of frame slots needed can be converted into actual memory requirements by considering the 80 bytes per frame used by the largest combination of AMR-WB+'s core and stereo rates.

The information about the frame buffer size is not always sufficient to determine when it is appropriate to start consuming frames from the interleaving buffer. There are two cases in which additional information is needed: first, when switching of the ISF occurs, and

second, when the interleaving pattern changes. The "int-delay" media type parameter is defined to convey this information. It allows a sender to indicate the minimal media time that needs to be present in the buffer before the decoder can start consuming frames from the buffer. Because the sender has full control over ISF changes and the interleaving pattern, it can calculate this value.

In certain cases (for example, if joining a multicast session with interleaving mid-session), a receiver may initially receive only part of the packets in the interleaving pattern. This initial partial reception (in frame sequence order) of frames can yield too few frames for acceptable quality from the audio decoding. This problem also arises when using encryption for access control, and the receiver does not have the previous key.

Although the AMR-WB+ is robust and thus tolerant to a high random frame erasure rate, it would have difficulties handling consecutive frame losses at startup. Thus, some special implementation considerations are described. In order to handle this type of startup efficiently, it must be noted that decoding is only possible to start at the beginning of a super-frame, and that holds true even if the first transport frame is indicated as lost. Secondly, decoding is only RECOMMENDED to start if at least 2 transport frames are available out of the 4 belonging to that super-frame.

After receiving a number of packets, in the worst case as many packets as the interleaving pattern covers, the previously described effects disappear and normal decoding is resumed.

Similar issues arise when a receiver leaves a session or has lost access to the stream. If the receiver leaves the session, this would be a minor issue since playout is normally stopped. It is also a minor issue for the case of lost access, since the AMR-WB+ error concealment will fade out the audio if massive consecutive losses are encountered.

The sender can avoid this type of problem in many sessions by starting and ending interleaving patterns correctly when risks of losses occur. One such example is a key-change done for access control to encrypted streams. If only some keys are provided to clients and there is a risk of their receiving content for which they do not have the key, it is recommended that interleaving patterns not

overlap key changes.

[4.5.](#) Implementation Considerations

An application implementing this payload format MUST understand all the payload parameters. Any mapping of the parameters to a signaling protocol MUST support all parameters. So an implementation of this payload format in an application using SDP is required to understand all the payload parameters in their SDP-mapped form. This requirement ensures that an implementation always can decide whether it is capable of communicating.

Both basic and interleaved mode SHALL be implemented. The implementation burden of both is rather small, and requiring both ensures interoperability. As the AMR-WB+ codec contains the full functionality of the AMR-WB codec, it is RECOMMENDED to also implement the payload format in [RFC 3267](#) [7] for the AMR-WB frame types when implementing this specification. Doing so makes interoperability with devices that only support AMR-WB more likely.

The switching of ISF, when combined with packet loss, could result in concealment using the wrong audio frame length. This can occur if packet losses result in lost frames directly after the point of ISF change. The packet loss would prevent the receiver from noticing the changed ISF and thereby conceal the lost transport frame with the previous ISF, instead of the new one. Although always later detectable, such an error results in frame boundary misalignment, which can cause audio distortions and problems with synchronization, as too many or too few audio samples were created. This problem can be mitigated in most cases by performing ISF recovery prior to concealment as outlined in [Section 4.5.1](#).

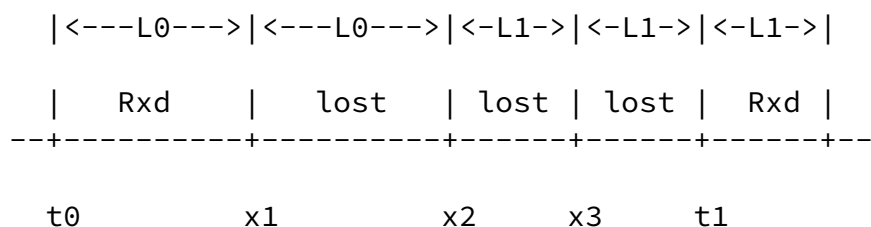
[4.5.1.](#) ISF Recovery in Case of Packet Loss

In case of packet loss, it is important that the AMR-WB+ decoder initiates a proper error concealment to replace the frames carried in the lost packet. A loss concealment algorithm requires a codec framing that matches the timestamps of the correctly received frames. Hence, it is necessary to recover the timestamps of the lost frames. Doing so is non-trivial because the codec frame length that is

associated with the ISF may have changed during the frame loss.

In the following, the recovery of the timestamp information of lost frames is illustrated by the means of an example. Two frames with timestamps t_0 and t_1 have been received properly, the first one being the last packet before the loss, and the latter one being the first packet after the loss period. The ISF values for these packets are isf_0 and isf_1 , respectively. The TFIs of these frames are t_{fi0} and t_{fi1} , respectively. The associated frame lengths (in timestamp ticks) are given as L_0 and L_1 , respectively. In this example three frames with timestamps $x_1 - x_3$ have been lost. The example further assumes that ISF changes once from isf_0 to isf_1 during the frame loss period, as shown in the figure below.

Since not all information required for the full recovery of the timestamps is generally known in the receiver, an algorithm is needed to estimate the ISF associated with the lost frames. Also, the number of lost frames needs to be recovered.



Example Algorithm:

```
Start:                                # check for frame loss
If (t0 + L0) == t1 Then goto End      # no frame loss

Step 1:                               # check case with no ISF change
If (isf0 != isf1) Then goto Step 2  # At least one ISF change
If (isFractional(t1 - t0)/L0) Then goto Step 3
                                     # More than 1 ISF change

Return recovered timestamps as
x(n) = t0 + n*L1 and associated ISF equal to isf0,
for 0 < n < (t1 - t0)/L0
goto End
```

```

Step 2:
Loop initialization: n := 4 - tfi0 mod 4
While n <= (t1-t0)/L0
    Evaluate m := (t1 - t0 - n*L0)/L1
    If (isInteger(m) AND ((tfi0+n+m) mod 4 == tfi1)) Then goto found;
    n := n+4
endloop
goto step 3                                     # More than 1 ISF change

```

```

found:
Return recovered timestamps and ISFs as
x(i) = t0 + i*L0 and associated ISF equal to isf0, for 0 < i <= n
x(i) = t0 + n*L0 + (i-n)*L1 and associated ISF equal to isf1,
for n < i <= n+m
goto End

```

Step 3:
 More than 1 ISF change has occurred. Since ISF changes can be assumed to be infrequent, such a situation occurs only if long sequences of frames are lost. In that case it is probably not useful to try to recover the timestamps of the lost frames. Rather, the AMR-WB+ decoder should be reset, and decoding should be resumed starting with the frame with timestamp t1.

End:

The above algorithm still does not solve the issue when the receiver buffer depth is shallower than the loss burst. In this kind of case, where the concealment must be done without any knowledge about future frames, the concealment may result in loss of frame boundary alignment. If that occurs, it may be necessary to reset and restart the codec to perform resynchronization.

[4.5.2.](#) Decoding Validation

If the receiver finds a mismatch between the size of a received payload and the size indicated by the ToC of the payload, the receiver SHOULD discard the packet. This is recommended because decoding a frame parsed from a payload based on erroneous ToC data could severely degrade the audio quality.

[5.](#) Congestion Control

The general congestion control considerations for transporting RTP data apply; see RTP [3] and any applicable RTP profile like AVP [9]. However, the multi-rate capability of AMR-WB+ audio coding provides a mechanism that may help to control congestion, since the bandwidth demand can be adjusted (within the limits of the codec) by selecting a different coding frame type or lower internal sampling rate.

The number of frames encapsulated in each RTP payload highly influences the overall bandwidth of the RTP stream due to header overhead constraints. Packetizing more frames in each RTP payload can reduce the number of packets sent and hence the header overhead, at the expense of increased delay and reduced error robustness.

If forward error correction (FEC) is used, the amount of FEC-induced redundancy needs to be regulated such that the use of FEC itself does not cause a congestion problem.

[6.](#) Security Considerations

RTP packets using the payload format defined in this specification are subject to the general security considerations discussed in RTP [3] and any applicable profile such as AVP [9] or SAVP [10]. As this format transports encoded audio, the main security issues include confidentiality, integrity protection, and data origin authentication of the audio itself. The payload format itself does not have any built-in security mechanisms. Any suitable external mechanisms, such as SRTP [10], MAY be used.

This payload format and the AMR-WB+ decoder do not exhibit any significant non-uniformity in the receiver-side computational complexity for packet processing, and thus are unlikely to pose a denial-of-service threat due to the receipt of pathological data.

[6.1.](#) Confidentiality

In order to ensure confidentiality of the encoded audio, all audio data bits **MUST** be encrypted. There is less need to encrypt the payload header or the table of contents since they only carry information about the frame type. This information could also be useful to a third party, for example, for quality monitoring.

The use of interleaving in conjunction with encryption can have a negative impact on confidentiality, for a short period of time. Consider the following packets (in brackets) containing frame numbers as indicated: {10, 14, 18}, {13, 17, 21}, {16, 20, 24} (a popular continuous diagonal interleaving pattern). The originator wishes to deny some participants the ability to hear material starting at time 16. Simply changing the key on the packet with the timestamp at or after 16, and denying that new key to those participants, does not achieve this; frames 17, 18, and 21 have been supplied in prior packets under the prior key, and error concealment may make the audio intelligible at least as far as frame 18 or 19, and possibly further.

[6.2](#). Authentication and Integrity

To authenticate the sender of the speech, an external mechanism **MUST** be used. It is **RECOMMENDED** that such a mechanism protects both the complete RTP header and the payload (speech and data bits).

Data tampering by a man-in-the-middle attacker could replace audio content and also result in erroneous depacketization/decoding that could lower the audio quality.

[7](#). Payload Format Parameters

This section defines the parameters that may be used to select features of the AMR-WB+ payload format. The parameters are defined as part of the media type registration for the AMR-WB+ audio codec. A mapping of the parameters into the Session Description Protocol (SDP) [[6](#)] is also provided for those applications that use SDP. Equivalent parameters could be defined elsewhere for use with control protocols that do not use MIME or SDP.

The data format and parameters are only specified for real-time transport in RTP.

7.1. Media Type Registration

The media type for the Extended Adaptive Multi-Rate Wideband (AMR-WB+) codec is allocated from the IETF tree, since AMR-WB+ is expected to be a widely used audio codec in general streaming applications.

Note: Parameters not listed below MUST be ignored by the receiver.

Media Type name: audio

Media subtype name: AMR-WB+

Required parameters:

None

Optional parameters:

channels: The maximum number of audio channels used by the audio frames. Permissible values are 1 (mono) or 2 (stereo). If no parameter is present, the maximum number of channels is 2 (stereo). Note: When set to 1, implicitly the stereo frame types cannot be used.

interleaving: Indicates that interleaved mode SHALL be used for the payload. The parameter specifies the number of transport frame slots required in a deinterleaving buffer (including the frame that is ready to be consumed). Its value is equal to one plus the maximum number of frames that precede any frame in transmission order and follow the frame in RTP timestamp order. The value MUST be greater than zero. If this parameter is not present, interleaved mode SHALL NOT be used.

int-delay: The minimal media time delay in RTP timestamp ticks that is needed in the deinterleaving buffer, i.e., the difference in RTP timestamp ticks between the earliest and latest audio frame present in the deinterleaving buffer.

ptime: See [Section 6 in RFC 2327](#) [6].

maxptime: See [Section 8 in RFC 3267](#) [7].

Restriction on Usage:

This type is only defined for transfer via RTP (STD 64).

[RFC 4352](#)

RTP Payload Format for AMR-WB+

January 2006

Encoding considerations:

An RTP payload according to this format is binary data and thus may need to be appropriately encoded in non-binary environments. However, as long as used within RTP, no encoding is necessary.

Security considerations:

See [Section 6 of RFC 4352](#).

Interoperability considerations:

To maintain interoperability with AMR-WB-capable end-points, in cases where negotiation is possible and the AMR-WB+ end-point supporting this format also supports [RFC 3267](#) for AMR-WB transport, an AMR-WB+ end-point SHOULD declare itself also as AMR-WB capable (i.e., supporting also "audio/AMR-WB" as specified in [RFC 3267](#)).

As the AMR-WB+ decoder is capable of performing stereo to mono conversions, all receivers of AMR-WB+ should be able to receive both stereo and mono, although the receiver is only capable of playout of mono signals.

Public specification:

[RFC 4352](#)

3GPP TS 26.290, see reference [1] of [RFC 4352](#)

Additional information:

This MIME type is not applicable for file storage. Instead, file storage of AMR-WB+ encoded audio is specified within the 3GPP-defined ISO-based multimedia file format defined in 3GPP TS 26.244; see reference [14] of [RFC 4352](#). This file format has the MIME types "audio/3GPP" or "video/3GPP" as defined by [RFC 3839](#) [15].

Person & email address to contact for further information:

magnus.westerlund@ericsson.com
ari.lakaniemi@nokia.com

Intended usage: COMMON.

It is expected that many IP-based streaming applications will use this type.

Change controller:

IETF Audio/Video Transport working group delegated from the IESG.

[7.2.](#) Mapping Media Type Parameters into SDP

The information carried in the media type specification has a specific mapping to fields in the Session Description Protocol (SDP) [\[6\]](#), which is commonly used to describe RTP sessions. When SDP is used to specify an RTP session using this RTP payload format, the mapping is as follows:

- The media type ("audio") is used in SDP "m=" as the media name.
- The media type (payload format name) is used in SDP "a=rtpmap" as the encoding name. The RTP clock rate in "a=rtpmap" SHALL be 72000 for AMR-WB+, and the encoding parameter number of channels MUST either be explicitly set to 1 or 2, or be omitted, implying the default value of 2.
- The parameters "ptime" and "maxptime" are placed in the SDP attributes "a=ptime" and "a=maxptime", respectively.
- Any remaining parameters are placed in the SDP "a=fmtp" attribute by copying them directly from the MIME media type string as a semicolon-separated list of parameter=value pairs.

[7.2.1.](#) Offer-Answer Model Considerations

To achieve good interoperability in an Offer-Answer [\[8\]](#) negotiation usage, the following considerations should be taken into account:

For negotiable offer/answer usage the following interpretation rules SHALL be applied:

- The "interleaving" parameter is symmetric, thus requiring that the answerer must also include it for the answer to an offered payload type that contains the parameter. However, the buffer space value is declarative in usage in unicast. For multicast usage, the same value in the response is required in order to accept the payload

type. For streams declared as `sendrecv` or `recvonly`: The receiver will accept reception of streams using the interleaved mode of the payload format. The value declares the amount of buffer space the receiver has available for the sender to utilize. For `sendonly` streams, the parameter indicates the desired configuration and amount of buffer space. An answerer is RECOMMENDED to respond using the offered value, if capable of using it.

- The "int-delay" parameter is declarative. For streams declared as `sendrecv` or `recvonly`, the value indicates the maximum initial delay the receiver will accept in the deinterleaving buffer. For `sendonly` streams, the value is the amount of media time the sender desires to use. The value SHOULD be copied into any response.
- The "channels" parameter is declarative. For "sendonly" streams, it indicates the desired channel usage, stereo and mono, or mono only. For "recvonly" and "sendrecv" streams, the parameter indicates what the receiver accepts to use. As any receiver will be capable of receiving stereo frame type and perform local mixing within the AMR-WB+ decoder, there is normally only one reason to restrict to mono only: to avoid spending bit-rate on data that are not utilized if the front-end is only capable of mono.
- The "ptime" parameter works as indicated by the offer/answer model [8]; "maxptime" SHALL be used in the same way.
- To maintain interoperability with AMR-WB in cases where negotiation is possible, an AMR-WB+ capable end-point that also implements the AMR-WB payload format [7] is RECOMMENDED to declare itself capable of AMR-WB as it is a subset of the AMR-WB+ codec.

In declarative usage, like SDP in RTSP [16] or SAP [17], the following interpretation of the parameters SHALL be done:

- The "interleaving" parameter, if present, configures the payload format in that mode, and the value indicates the number of frames that the deinterleaving buffer is required to support to be able

to handle this session correctly.

- The "int-delay" parameter indicates the initial buffering delay required to receive this stream correctly.
- The "channels" parameter indicates if the content being transmitted can contain either both stereo and mono rates, or only mono.
- All other parameters indicate values that are being used by the sending entity.

[7.2.2.](#) Examples

One example of an SDP session description utilizing AMR-WB+ mono and stereo encoding follows.

```
m=audio 49120 RTP/AVP 99
a=rtpmap:99 AMR-WB+/72000/2
a=fmtp:99 interleaving=30; int-delay=86400
a=maxptime:100
```

Note that the payload format (encoding) names are commonly shown in uppercase. Media subtypes are commonly shown in lowercase. These names are case-insensitive in both places. Similarly, parameter names are case-insensitive both in MIME types and in the default mapping to the SDP a=fmtp attribute.

[8.](#) IANA Considerations

The IANA has registered one new MIME subtype (audio/amr-wb+); see [Section 7](#).

[9.](#) Contributors

Daniel Enstrom has contributed in writing the codec introduction section. Stefan Bruhn has contributed by writing the ISF recovery algorithm.

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Any 3GPP document can be downloaded from the 3GPP webserver, "http://www.3gpp.org/", see specifications.

Johan Sjöberg
Ericsson Research
Ericsson AB
SE-164 80 Stockholm
SWEDEN

Phone: +46 8 7190000
EMail: Johan.Sjoberg@ericsson.com

Magnus Westerlund
Ericsson Research
Ericsson AB
SE-164 80 Stockholm
SWEDEN

Phone: +46 8 7190000
EMail: Magnus.Westerlund@ericsson.com

Ari Lakaniemi
Nokia Research Center
P.O. Box 407
FIN-00045 Nokia Group
FINLAND

Phone: +358-71-8008000
EMail: ari.lakaniemi@nokia.com

Stephan Wenger
Nokia Corporation
P.O. Box 100
FIN-33721 Tampere
FINLAND

Phone: +358-50-486-0637
EMail: Stephan.Wenger@nokia.com

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