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RTP Payload Format and File Storage Format for the
Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB)
Audio Codecs

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 to be used for Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) encoded speech signals. The payload format is designed to be able to interoperate with existing AMR and AMR-WB transport formats on non-IP networks. In addition, a file format is specified for transport of AMR and AMR-WB speech data in storage mode applications such as email. Two separate media type registrations are included, one for AMR and one for AMR-WB, specifying use of both the RTP payload format and the storage format. This document obsoletes RFC 3267.

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1. Introduction

This document obsoletes RFC 3267 and extends that specification with offer/answer rules. See Section 10 for the changes made to this format in relation to RFC 3267.

This document specifies the payload format for packetization of AMR and AMR-WB encoded speech signals into the Real-time Transport Protocol (RTP) [8]. The payload format supports transmission of multiple channels, multiple frames per payload, the use of fast codec mode adaptation, robustness against packet loss and bit errors, and interoperation with existing AMR and AMR-WB transport formats on non-IP networks, as described in Section 3.

The payload format itself is specified in Section 4. A related file format is specified in Section 5 for transport of AMR and AMR-WB speech data in storage mode applications such as email. In Section 8, two separate media type registrations are provided, one for AMR and one for AMR-WB.

Even though this RTP payload format definition supports the transport of both AMR and AMR-WB speech, it is important to remember that AMR and AMR-WB are two different codecs and they are always handled as different payload types in RTP.

2. Conventions and Acronyms

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 [5].

The following acronyms are used in this document:

3GPP	- the Third Generation Partnership Project
AMR	- Adaptive Multi-Rate (Codec)
AMR-WB	- Adaptive Multi-Rate Wideband (Codec)
CMR	- Codec Mode Request
CN	- Comfort Noise
DTX	- Discontinuous Transmission
ETSI	- European Telecommunications Standards Institute
FEC	- Forward Error Correction
SCR	- Source Controlled Rate Operation
SID	- Silence Indicator (the frames containing only CN parameters)
VAD	- Voice Activity Detection
UED	- Unequal Error Detection
UEP	- Unequal Error Protection

The term "frame-block" is used in this document to describe the time-synchronized set of speech frames in a multi-channel AMR or AMR-WB session. In particular, in an N-channel session, a frame-block will contain N speech frames, one from each of the channels, and all N speech frames represents exactly the same time period.

The byte order used in this document is network byte order, i.e., the most significant byte first. The bit order is also the most significant bit first. This is presented in all figures as having the most significant bit leftmost on a line and with the lowest number. Some bit fields may wrap over multiple lines in which cases the bits on the first line are more significant than the bits on the next line.

3. Background on AMR/AMR-WB and Design Principles

AMR and AMR-WB were originally designed for circuit-switched mobile radio systems. Due to their flexibility and robustness, they are also suitable for other real-time speech communication services over packet-switched networks such as the Internet.

Because of the flexibility of these codecs, the behavior in a particular application is controlled by several parameters that select options or specify the acceptable values for a variable. These options and variables are described in general terms at appropriate points in the text of this specification as parameters to be established through out-of-band means. In Section 8, all of the parameters are specified in the form of media subtype registrations for the AMR and AMR-WB encodings. 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 8.3 provides a mapping of the parameters into the Session Description Protocol (SDP) [11] for those applications that use SDP.

3.1. The Adaptive Multi-Rate (AMR) Speech Codec

The AMR codec was originally developed and standardized by the European Telecommunications Standards Institute (ETSI) for GSM cellular systems. It is now chosen by the Third Generation Partnership Project (3GPP) as the mandatory codec for third generation (3G) cellular systems [1].

The AMR codec is a multi-mode codec that supports eight narrow band speech encoding modes with bit rates between 4.75 and 12.2 kbps. The sampling frequency used in AMR is 8000 Hz and the speech encoding is performed on 20 ms speech frames. Therefore, each encoded AMR speech frame represents 160 samples of the original speech.

Among the eight AMR encoding modes, three are already separately adopted as standards of their own. Particularly, the 6.7 kbps mode is adopted as PDC-EFR [18], the 7.4 kbps mode as IS-641 codec in TDMA [17], and the 12.2 kbps mode as GSM-EFR [16].

3.2. The Adaptive Multi-Rate Wideband (AMR-WB) Speech Codec

The Adaptive Multi-Rate Wideband (AMR-WB) speech codec [3] was originally developed by 3GPP to be used in GSM and 3G cellular systems.

Similar to AMR, the AMR-WB codec is also a multi-mode speech codec. AMR-WB supports nine wide band speech coding modes with respective bit rates ranging from 6.6 to 23.85 kbps. The sampling frequency used in AMR-WB is 16000 Hz and the speech processing is performed on 20 ms frames. This means that each AMR-WB encoded frame represents 320 speech samples.

3.3. Multi-Rate Encoding and Mode Adaptation

The multi-rate encoding (i.e., multi-mode) capability of AMR and AMR-WB is designed for preserving high speech quality under a wide range of transmission conditions.

With AMR or AMR-WB, mobile radio systems are able to use available bandwidth as effectively as possible. For example, in GSM it is possible to dynamically adjust the speech encoding rate during a session so as to continuously adapt to the varying transmission conditions by dividing the fixed overall bandwidth between speech data and error protective coding. This enables the best possible trade-off between speech compression rate and error tolerance. To perform mode adaptation, the decoder (speech receiver) needs to signal the encoder (speech sender) the new mode it prefers. This mode change signal is called Codec Mode Request or CMR.

Since in most sessions speech is sent in both directions between the two ends, the mode requests from the decoder at one end to the encoder at the other end are piggy-backed over the speech frames in the reverse direction. In other words, there is no out-of-band signaling needed for sending CMRs.

Every AMR or AMR-WB codec implementation is required to support all the respective speech coding modes defined by the codec and must be able to handle mode switching to any of the modes at any time. However, some transport systems may impose limitations in the number of modes supported and how often the mode can change due to bandwidth

limitations or other constraints. For this reason, the decoder is allowed to indicate its acceptance of a particular mode or a subset of the defined modes for the session using out-of-band means.

For example, the GSM radio link can only use a subset of at most four different modes in a given session. This subset can be any combination of the eight AMR modes for an AMR session or any combination of the nine AMR-WB modes for an AMR-WB session.

Moreover, for better interoperability with GSM through a gateway, the decoder is allowed to use out-of-band means to set the minimum number of frames between two mode changes and to limit the mode change among neighboring modes only.

Section 8 specifies a set of media type parameters that may be used to signal these mode adaptation controls at session setup.

3.4. Voice Activity Detection and Discontinuous Transmission

Both codecs support voice activity detection (VAD) and generation of comfort noise (CN) parameters during silence periods. Hence, the codecs have the option to reduce 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 or AMR-WB frames containing CN parameters are called Silence Indicator (SID) frames. See more details about VAD and DTX functionality in [9] and [10].

3.5. Support for Multi-Channel Session

Both the RTP payload format and the storage format defined in this document support multi-channel audio content (e.g., a stereophonic speech session).

Although AMR and AMR-WB codecs themselves do not support encoding of multi-channel audio content into a single bit stream, they can be used to separately encode and decode each of the individual channels.

To transport (or store) the separately encoded multi-channel content, the speech frames for all channels that are framed and encoded for the same 20 ms periods are logically collected in a frame-block.

At the session setup, out-of-band signaling must be used to indicate the number of channels in the session, and the order of the speech frames from different channels in each frame-block. When using SDP for signaling, the number of channels is specified in the `rtptime` attribute and the order of channels carried in each frame-block is

implied by the number of channels as specified in Section 4.1 in [12].

3.6. Unequal Bit-Error Detection and Protection

The speech bits encoded in each AMR or AMR-WB frame have different perceptual sensitivity to bit errors. This property has been exploited in cellular systems to achieve better voice quality by using unequal error protection and detection (UEP and UED) mechanisms.

The UEP/UED mechanisms focus the protection and detection of corrupted bits to the perceptually most sensitive bits in an AMR or AMR-WB frame. In particular, speech bits in an AMR or AMR-WB frame are divided into class A, B, and C, where bits in class A are the most sensitive and bits in class C the least sensitive (see Table 1 below for AMR and [4] for AMR-WB). An AMR or AMR-WB frame is only declared damaged if there are bit errors found in the most sensitive bits, i.e., the class A bits. On the other hand, it is acceptable to have some bit errors in the other bits, i.e., class B and C bits.

Index	Mode	Class A bits	Total speech bits
0	AMR 4.75	42	95
1	AMR 5.15	49	103
2	AMR 5.9	55	118
3	AMR 6.7	58	134
4	AMR 7.4	61	148
5	AMR 7.95	75	159
6	AMR 10.2	65	204
7	AMR 12.2	81	244
8	AMR SID	39	39

Table 1. The number of class A bits for the AMR codec

Moreover, a damaged frame is still useful for error concealment at the decoder since some of the less sensitive bits can still be used. This approach can improve the speech quality compared to discarding the damaged frame.

3.6.1. Applying UEP and UED in an IP Network

To take full advantage of the bit-error robustness of the AMR and AMR-WB codec, the RTP payload format is designed to facilitate UEP/UED in an IP network. It should be noted however that the utilization of UEP and UED discussed below is OPTIONAL.

UEP/UED in an IP network can be achieved by detecting bit errors in class A bits and tolerating bit errors in class B/C bits of the AMR or AMR-WB frame(s) in each RTP payload.

Link-layer protocols exist that do not discard packets containing bit errors, e.g., SLIP and some wireless links. With the Internet traffic pattern shifting towards a more multimedia-centric one, more link layers of such nature may emerge in the future. With transport layer support for partial checksums (for example, those supported by UDP-Lite [19]), bit error tolerant AMR and AMR-WB traffic could achieve better performance over these types of links. The relationship between UDP-Lite's partial checksum at the transport layer and the checksum coverage provided by the link-layer frame is described in UDP-Lite specification [19].

There are at least two basic approaches for carrying AMR and AMR-WB traffic over bit error tolerant IP networks:

- a) Utilizing a partial checksum to cover the IP, transport protocol (e.g., UDP-Lite), RTP and payload headers, and the most important speech bits of the payload. The IP, UDP and RTP headers need to be protected, and it is recommended that at least all class A bits are covered by the checksum.
- b) Utilizing a partial checksum to only cover the IP, transport protocol, RTP and payload headers, but an AMR or AMR-WB frame CRC to cover the class A bits of each speech frame in the RTP payload.

In either approach, at least part of the class B/C bits are left without error-check and thus bit error tolerance is achieved.

Note, it is still important that the network designer pays attention to the class B and C residual bit error rate. Though less sensitive to errors than class A bits, class B and C bits are not insignificant, and undetected errors in these bits cause degradation in speech quality. An example of residual error rates considered acceptable for AMR in the Universal Mobile Telecommunications System (UMTS) can be found in [24] and for AMR-WB in [25].

The application interface to the UEP/UED transport protocol (e.g., UDP-Lite) may not provide any control over the link error rate, especially in a gateway scenario. Therefore, it is incumbent upon the designer of a node with a link interface of this type to choose a residual bit error rate that is low enough to support applications such as AMR encoding when transmitting packets of a UEP/UED transport protocol.

Approach 1 is bit efficient, flexible and simple, but comes with two disadvantages, namely, a) bit errors in protected speech bits will cause the payload to be discarded, and b) when transporting multiple AMR or AMR-WB frames in a RTP payload, there is the possibility that a single bit error in protected bits will cause all the frames to be discarded.

These disadvantages can be avoided, if needed, with some overhead in the form of a frame-wise CRC (Approach 2). In problem a), the CRC makes it possible to detect bit errors in class A bits and use the frame for error concealment, which gives a small improvement in speech quality. For b), when transporting multiple frames in a payload, the CRCs remove the possibility that a single bit error in a class A bit will cause all the frames to be discarded. Avoiding that improves the speech quality when transporting multiple AMR or AMR-WB frames over links subject to bit errors.

The choice between the above two approaches must be made based on the available bandwidth, and the desired tolerance to bit errors. Neither solution is appropriate for all cases. Section 8 defines parameters that may be used at session setup to choose between these approaches.

3.7. Robustness against Packet Loss

The payload format supports several means, including forward error correction (FEC) and frame interleaving, to increase robustness against packet loss.

3.7.1. Use of Forward Error Correction (FEC)

The simple scheme of repetition of previously sent data is one way of achieving FEC. Another possible scheme which is more bandwidth efficient is to use payload-external FEC, e.g., RFC 2733 [23], which generates extra packets containing repair data. The whole payload can also be sorted in sensitivity order to support external FEC schemes using UEP. There is also a work in progress on a generic version of such a scheme [22] that can be applied to AMR or AMR-WB payload transport.

With AMR or AMR-WB, it is possible to use the multi-rate capability of the codec to send redundant copies of a frame using either the same mode or another mode, e.g., one with lower bandwidth. We describe such a scheme next.

receiver reports. A sender should not base selection of FEC on the CMR, as this parameter most probably was set based on non-IP information, e.g., radio link performance measures. The sender is also responsible for avoiding congestion, which may be exacerbated by redundancy (see Section 6 for more details).

3.7.2. Use of Frame Interleaving

To decrease protocol overhead, the payload design allows several speech frame-blocks to be encapsulated into a single RTP packet. One of the drawbacks of such an approach is that packet loss can cause loss of several consecutive speech frame-blocks, which usually causes clearly audible distortion in the reconstructed speech. Interleaving of frame-blocks can improve the speech quality in such cases by distributing the consecutive losses into a series of single frame-block losses. However, interleaving and bundling several frame-blocks per payload will also increase end-to-end delay and is therefore not appropriate for all types of applications. Streaming applications will most likely be able to exploit interleaving to improve speech quality in lossy transmission conditions.

This payload design supports the use of frame interleaving as an option. For the encoder (speech sender) to use frame interleaving in its outbound RTP packets for a given session, the decoder (speech receiver) needs to indicate its support via out-of-band means (see Section 8).

3.8. Bandwidth-Efficient or Octet-Aligned Mode

For a given session, the payload format can be either bandwidth efficient or octet aligned, depending on the mode of operation that is established for the session via out-of-band means.

In the octet-aligned format, all the fields in a payload, including payload header, table of contents entries, and speech frames themselves, are individually aligned to octet boundaries to make implementations efficient. In the bandwidth-efficient format, only the full payload is octet aligned, so fewer padding bits are added.

Note, octet alignment of a field or payload means that the last octet is padded with zeroes in the least significant bits to fill the octet. Also note that this padding is separate from padding indicated by the P bit in the RTP header.

Between the two operation modes, only the octet-aligned mode has the capability to use the robust sorting, interleaving, and frame CRC to make the speech transport more robust to packet loss and bit errors.

3.9. AMR or AMR-WB Speech over IP Scenarios

The primary scenario for this payload format is IP end-to-end between two terminals, as shown in Figure 2. This payload format is expected to be useful for both conversational and streaming services.

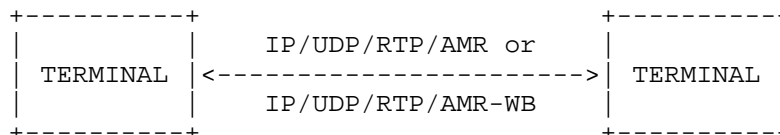


Figure 2: IP terminal to IP terminal scenario

A conversational service puts requirements on the payload format. Low delay is one very important factor, i.e., few speech frame-blocks per payload packet. Low overhead is also required when the payload format traverses low bandwidth links, especially as the frequency of packets will be high. For low bandwidth links, it is also an advantage to support UED, which allows a link provider to reduce delay and packet loss, or to reduce the utilization of link resources.

A streaming service has less strict real-time requirements and therefore can use a larger number of frame-blocks per packet than a conversational service. This reduces the overhead from IP, UDP, and RTP headers. However, including several frame-blocks per packet makes the transmission more vulnerable to packet loss, so interleaving may be used to reduce the effect that packet loss will have on speech quality. A streaming server handling a large number of clients also needs a payload format that requires as few resources as possible when doing packetization. The octet-aligned and interleaving modes require the least amount of resources, while CRC, robust sorting, and bandwidth-efficient modes have higher demands.

Another scenario is when AMR or AMR-WB encoded speech is transmitted from a non-IP system (e.g., a GSM or 3GPP UMTS network) to an IP/UDP/RTP VoIP terminal, and/or vice versa, as depicted in Figure 3.

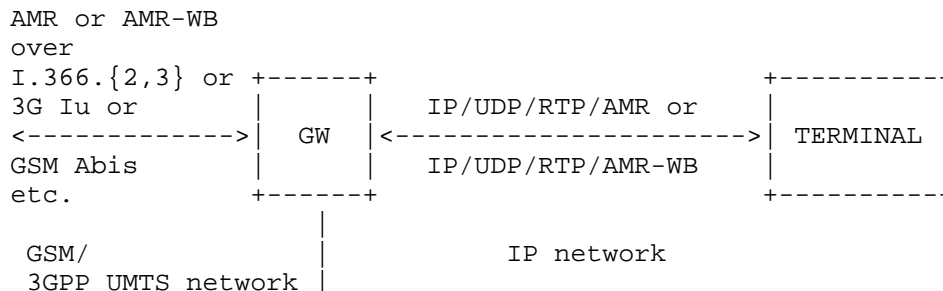


Figure 3: GW to VoIP terminal scenario

In such a case, it is likely that the AMR or AMR-WB frame is packetized in a different way in the non-IP network and will need to be re-packetized into RTP at the gateway. Also, speech frames from the non-IP network may come with some UEP/UED information (e.g., a frame quality indicator) that will need to be preserved and forwarded on to the decoder along with the speech bits. This is specified in Section 4.3.2.

AMR's capability to do fast mode switching is exploited in some non-IP networks to optimize speech quality. To preserve this functionality in scenarios including a gateway to an IP network, a codec mode request (CMR) field is needed. The gateway will be responsible for forwarding the CMR between the non-IP and IP parts in both directions. The IP terminal should follow the CMR forwarded by the gateway to optimize speech quality going to the non-IP decoder. The mode control algorithm in the gateway must accommodate the delay imposed by the IP network on the IP terminal's response to CMR.

The IP terminal should not set the CMR (see Section 4.3.1), but the gateway can set the CMR value on frames going toward the encoder in the non-IP part to optimize speech quality from that encoder to the gateway. The gateway can alternatively set a lower CMR value, if desired, as one means to control congestion on the IP network.

A third likely scenario is that IP/UDP/RTP is used as transport between two non-IP systems, i.e., IP is originated and terminated in gateways on both sides of the IP transport, as illustrated in Figure 4 below.

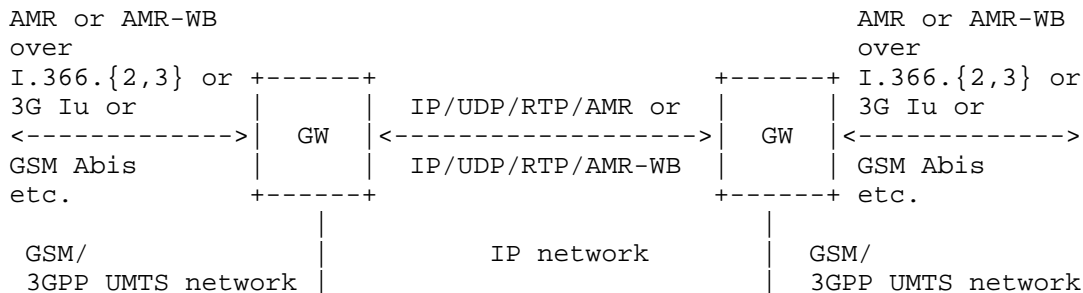


Figure 4: GW to GW scenario

This scenario requires the same mechanisms for preserving UED/UEP and CMR information as in the single gateway scenario. In addition, the CMR value may be set in packets received by the gateways on the IP network side. The gateway should forward to the non-IP side a CMR value that is the minimum of three values:

- the CMR value it receives on the IP side;
- the CMR value it calculates based on its reception quality on the non-IP side; and
- a CMR value it may choose for congestion control of transmission on the IP side.

The details of the control algorithm are left to the implementation.

4. AMR and AMR-WB RTP Payload Formats

The AMR and AMR-WB payload formats have identical structure, so they are specified together. The only differences are in the types of codec frames contained in the payload. The payload format consists of the RTP header, payload header, and payload data.

4.1. RTP Header Usage

The format of the RTP header is specified in [8]. 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-block in the packet. The timestamp clock frequency is the same as the sampling frequency, so the timestamp unit is in samples.

The duration of one speech frame-block is 20 ms for both AMR and AMR-WB. For AMR, the sampling frequency is 8 kHz, corresponding to 160 encoded speech samples per frame from each channel. For AMR-WB, the sampling frequency is 16 kHz, corresponding to 320 samples per frame from each channel. Thus, the timestamp is increased by 160 for AMR and 320 for AMR-WB for each consecutive frame-block.

A packet may contain multiple frame-blocks of encoded speech or comfort noise parameters. If interleaving is employed, the frame-blocks encapsulated into a payload are picked according to the interleaving rules as defined in Section 4.4.1. Otherwise, each packet covers a period of one or more contiguous 20 ms frame-block intervals. In case the data from all the channels for a particular frame-block in the period is missing (for example, at a gateway from some other transport format), it is possible to indicate that no data is present for that frame-block rather than breaking a multi-frame-block packet into two, as explained in Section 4.3.2.

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 speech frame multiple times, in exact duplicates, in different AMR rate modes, or with data present in one packet and not present in another. If multiple versions of the same speech frame are received, it is RECOMMENDED that the mode with the highest rate be used by the speech decoder. A given frame MUST NOT be encoded as speech in one packet and comfort noise parameters in another.

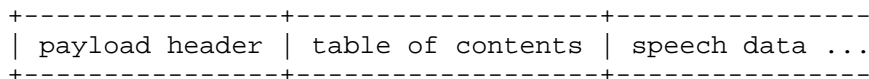
The payload length is always made an integral number of octets by padding with zero bits if necessary. If additional padding is required to bring the payload length to a larger multiple of octets or for some other purpose, then the P bit in the RTP in the header may be set and padding appended as specified in [8].

The RTP header marker bit (M) SHALL be set to 1 if the first frame-block carried in the packet contains a speech frame which is the first in a talkspurt. For all other packets the marker bit SHALL be set to zero (M=0).

The assignment of an RTP payload type for this new packet format is outside the scope of this document, and will not be specified here. It is expected that the RTP profile under which this payload format is being used will assign a payload type for this encoding or specify that the payload type is to be bound dynamically.

4.2. Payload Structure

The complete payload consists of a payload header, a payload table of contents, and speech data representing one or more speech frame-blocks. The following diagram shows the general payload format layout:



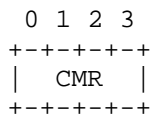
Payloads containing more than one speech frame-block are called compound payloads.

The following sections describe the variations taken by the payload format depending on whether the AMR session is set up to use the bandwidth-efficient mode or octet-aligned mode and any of the OPTIONAL functions for robust sorting, interleaving, and frame CRCs. Implementations SHOULD support both bandwidth-efficient and octet-aligned operation to increase interoperability.

4.3. Bandwidth-Efficient Mode

4.3.1. The Payload Header

In bandwidth-efficient mode, the payload header simply consists of a 4-bit codec mode request:



CMR (4 bits): Indicates a codec mode request sent to the speech encoder at the site of the receiver of this payload. The value of the CMR field is set to the frame type index of the corresponding speech mode being requested. The frame type index may be 0-7 for AMR, as defined in Table 1a in [2], or 0-8 for AMR-WB, as defined in Table 1a in [4]. CMR value 15 indicates that no mode request is present, and other values are for future use.

The codec mode request received in the CMR field is valid until the next codec mode request is received, i.e., a newly received CMR value corresponding to a speech mode, or NO_DATA overrides the previously received CMR value corresponding to a speech mode or NO_DATA. Therefore, if a terminal continuously wishes to receive frames in the

same mode X, it needs to set CMR=X for all its outbound payloads, and if a terminal has no preference in which mode to receive, it SHOULD set CMR=15 in all its outbound payloads.

If receiving a payload with a CMR value that is not a speech mode or NO_DATA, the CMR MUST be ignored by the receiver.

In a multi-channel session, the codec mode request SHOULD be interpreted by the receiver of the payload as the desired encoding mode for all the channels in the session.

An IP end-point SHOULD NOT set the codec mode request based on packet losses or other congestion indications, for several reasons:

- The other end of the IP path may be a gateway to a non-IP network (such as a radio link) that needs to set the CMR field to optimize performance on that network.
- Congestion on the IP network is managed by the IP sender, in this case, at the other end of the IP path. Feedback about congestion SHOULD be provided to that IP sender through RTCP or other means, and then the sender can choose to avoid congestion using the most appropriate mechanism. That may include adjusting the codec mode, but also includes adjusting the level of redundancy or number of frames per packet.

The encoder SHOULD follow a received codec mode request, but MAY change to a lower-numbered mode if it so chooses, for example, to control congestion.

The CMR field MUST be set to 15 for packets sent to a multicast group. The encoder in the speech sender SHOULD ignore codec mode requests when sending speech to a multicast session but MAY use RTCP feedback information as a hint that a codec mode change is needed.

The codec mode selection MAY be restricted by a session parameter to a subset of the available modes. If so, the requested mode MUST be among the signalled subset (see Section 8). If the received CMR value is outside the signalled subset of modes, it MUST be ignored.

4.3.2. The Payload Table of Contents

The table of contents (ToC) consists of a list of ToC entries, each representing a speech frame.

In bandwidth-efficient mode, a ToC entry takes the following format:

```

 0 1 2 3 4 5
+-----+
|F|  FT  |Q|
+-----+

```

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

FT (4 bits): Frame type index, indicating either the AMR or AMR-WB speech coding mode or comfort noise (SID) mode of the corresponding frame carried in this payload.

The value of FT is defined in Table 1a in [2] for AMR and in Table 1a in [4] for AMR-WB. FT=14 (SPEECH_LOST, only available for AMR-WB) and FT=15 (NO_DATA) are used to indicate frames that are either lost or not being transmitted in this payload, respectively.

NO_DATA (FT=15) frame could mean either that no data for that frame has been produced by the speech encoder or that no data for that frame is transmitted in the current payload (i.e., valid data for that frame could be sent in either an earlier or later packet).

If receiving a ToC entry with a FT value in the range 9-14 for AMR or 10-13 for AMR-WB, the whole packet SHOULD be discarded. This is to avoid the loss of data synchronization in the depacketization process, which can result in a huge degradation in speech quality.

Note that packets containing only NO_DATA frames SHOULD NOT be transmitted in any payload format configuration, except in the case of interleaving. Also, frame-blocks containing only NO_DATA frames at the end of a packet SHOULD NOT be transmitted in any payload format configuration, except in the case of interleaving. The AMR SCR/DTX is described in [6] and AMR-WB SCR/DTX in [7].

The extra comfort noise frame types specified in table 1a in [2] (i.e., GSM-EFR CN, IS-641 CN, and PDC-EFR CN) MUST NOT be used in this payload format because the standardized AMR codec is only required to implement the general AMR SID frame type and not those that are native to the incorporated encodings.

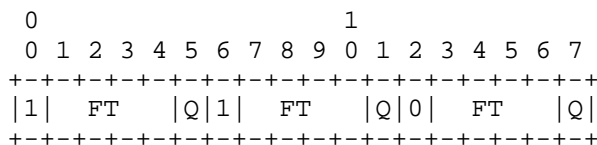
Q (1 bit): Frame quality indicator. If set to 0, indicates the corresponding frame is severely damaged, and the receiver should set the RX_TYPE (see [6]) to either SPEECH_BAD or SID_BAD depending on the frame type (FT).

The frame quality indicator is included for interoperability with the ATM payload format described in ITU-T I.366.2, the UMTS Iu interface [20], as well as other transport formats. The frame quality indicator enables damaged frames to be forwarded to the speech decoder for error concealment. This can improve the speech quality more than dropping the damaged frames. See Section 4.4.2.1 for more details.

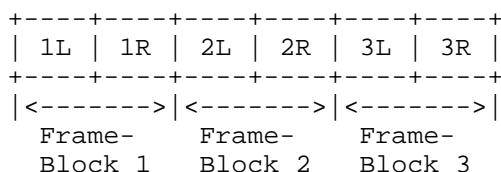
For multi-channel sessions, the ToC entries of all frames from a frame-block are placed in the ToC in consecutive order as defined in Section 4.1 in [12]. When multiple frame-blocks are present in a packet in bandwidth-efficient mode, they will be placed in the packet in order of their creation time.

Therefore, with N channels and K speech frame-blocks in a packet, there MUST be N*K entries in the ToC, and the first N entries will be from the first frame-block, the second N entries will be from the second frame-block, and so on.

The following figure shows an example of a ToC of three entries in a single-channel session using bandwidth-efficient mode.



Below is an example of how the ToC entries will appear in the ToC of a packet carrying three consecutive frame-blocks in a session with two channels (L and R).



4.3.3. Speech Data

Speech data of a payload contains zero or more speech frames or comfort noise frames, as described in the ToC of the payload.

Note, for ToC entries with FT=14 or 15, there will be no corresponding speech frame present in the speech data.

Each speech frame represents 20 ms of speech encoded with the mode indicated in the FT field of the corresponding ToC entry. The length of the speech frame is implicitly defined by the mode indicated in the FT field. The order and numbering notation of the bits are as specified for Interface Format 1 (IF1) in [2] for AMR and [4] for AMR-WB. As specified there, the bits of speech frames have been rearranged in order of decreasing sensitivity, while the bits of comfort noise frames are in the order produced by the encoder. The resulting bit sequence for a frame of length K bits is denoted $d(0), d(1), \dots, d(K-1)$.

4.3.4. Algorithm for Forming the Payload

The complete RTP payload in bandwidth-efficient mode is formed by packing bits from the payload header, table of contents, and speech frames in order (as defined by their corresponding ToC entries in the ToC list), and to bring the payload to octet alignment, 0 to 7 padding bits. Padding bits MUST be set to zero and MUST be ignored on reception. They are packed contiguously into octets beginning with the most significant bits of the fields and the octets.

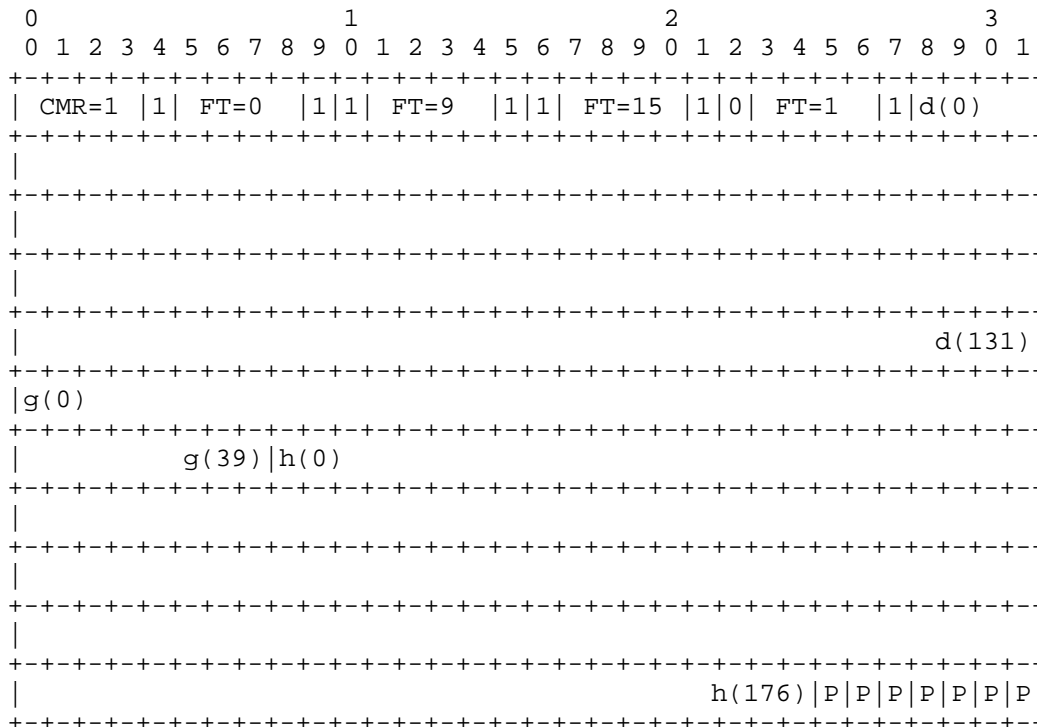
To be precise, the four-bit payload header is packed into the first octet of the payload with bit 0 of the payload header in the most significant bit of the octet. The four most significant bits (numbered 0-3) of the first ToC entry are packed into the least significant bits of the octet, ending with bit 3 in the least significant bit. Packing continues in the second octet with bit 4 of the first ToC entry in the most significant bit of the octet. If more than one frame is contained in the payload, then packing continues with the second and successive ToC entries. Bit 0 of the first data frame follows immediately after the last ToC bit, proceeding through all the bits of the frame in numerical order. Bits from any successive frames follow contiguously in numerical order for each frame and in consecutive order of the frames.

If speech data is missing for one or more speech frame within the sequence, because of, for example, DTX, a ToC entry with FT set to NO_DATA SHALL be included in the ToC for each of the missing frames, but no data bits are included in the payload for the missing frame (see Section 4.3.5.2 for an example).

4.3.5. Payload Examples

4.3.5.1. Single-Channel Payload Carrying a Single Frame

The following diagram shows a bandwidth-efficient AMR payload from a single-channel session carrying a single speech frame-block.



4.3.5.3. Multi-Channel Payload Carrying Multiple Frames

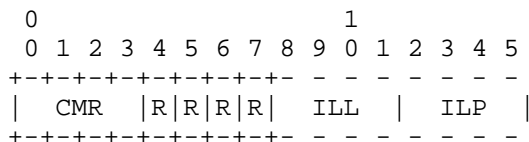
The following diagram shows a two-channel payload carrying 3 frame-blocks, i.e., the payload will contain 6 speech frames.

In the payload, all speech frames contain the same mode 7.4 kbps (FT=4) and are not damaged at IP origin. The CMR is set to 15, i.e., no specific mode is requested. The two channels are defined as left (L) and right (R) in that order. The encoded speech bits is designated dXY(0).. dXY(K-1), where X = block number, Y = channel, and K is the number of speech bits for that mode. Exemplifying this, for frame-block 1 of the left channel, the encoded bits are designated as d1L(0) to d1L(147).

4.4. Octet-Aligned Mode

4.4.1. The Payload Header

In octet-aligned mode, the payload header consists of a 4-bit CMR, 4 reserved bits, and optionally, an 8-bit interleaving header, as shown below:



CMR (4 bits): same as defined in Section 4.3.1.

R: is a reserved bit that MUST be set to zero. All R bits MUST be ignored by the receiver.

ILL (4 bits, unsigned integer): This is an OPTIONAL field that is present only if interleaving is signalled out-of-band for the session. ILL=L indicates to the receiver that the interleaving length is L+1, in number of frame-blocks.

ILP (4 bits, unsigned integer): This is an OPTIONAL field that is present only if interleaving is signalled. ILP MUST take a value between 0 and ILL, inclusive, indicating the interleaving index for frame-blocks in this payload in the interleaving group. If the value of ILP is found greater than ILL, the payload SHOULD be discarded.

ILL and ILP fields MUST be present in each packet in a session if interleaving is signalled for the session. Interleaving MUST be performed on a frame-block basis (i.e., NOT on a frame basis) in a multi-channel session.

The following example illustrates the arrangement of speech frame-blocks in an interleaving group during an interleaving session. Here we assume ILL=L for the interleaving group that starts at speech frame-block n. We also assume that the first payload packet of the interleaving group is s, and the number of speech frame-blocks carried in each payload is N. Then we will have:

Payload s (the first packet of this interleaving group):

ILL=L, ILP=0,
Carry frame-blocks: $n, n+(L+1), n+2*(L+1), \dots, n+(N-1)*(L+1)$

Payload $s+1$ (the second packet of this interleaving group):

ILL=L, ILP=1,
frame-blocks: $n+1, n+1+(L+1), n+1+2*(L+1), \dots, n+1+(N-1)*(L+1)$
...

Payload $s+L$ (the last packet of this interleaving group):

ILL=L, ILP=L,
frame-blocks: $n+L, n+L+(L+1), n+L+2*(L+1), \dots, n+L+(N-1)*(L+1)$

The next interleaving group will start at frame-block $n+N*(L+1)$.

There will be no interleaving effect unless the number of frame-blocks per packet (N) is at least 2. Moreover, the number of frame-blocks per payload (N) and the value of ILL MUST NOT be changed inside an interleaving group. In other words, all payloads in an interleaving group MUST have the same ILL and MUST contain the same number of speech frame-blocks.

The sender of the payload MUST only apply interleaving if the receiver has signalled its use through out-of-band means. Since interleaving will increase buffering requirements at the receiver, the receiver uses media type parameter "interleaving=I" to set the maximum number of frame-blocks allowed in an interleaving group to I .

When performing interleaving, the sender MUST use a proper number of frame-blocks per payload (N) and ILL so that the resulting size of an interleaving group is less or equal to I , that is, $N*(L+1) \leq I$.

4.4.2. The Payload Table of Contents and Frame CRCs

The table of contents (ToC) in octet-aligned mode consists of a list of ToC entries where each entry corresponds to a speech frame carried in the payload and, optionally, a list of speech frame CRCs. That is, the ToC is as follows:

```
+-----+
| list of ToC entries |
+-----+
| list of frame CRCs | (optional)
- - - - -
```

Note, for ToC entries with FT=14 or 15, there will be no corresponding speech frame or frame CRC present in the payload.

The list of ToC entries is organized in the same way as described for bandwidth-efficient mode in 4.3.2, with the following exception: when interleaving is used, the frame-blocks in the ToC will almost never be placed consecutively in time. Instead, the presence and order of the frame-blocks in a packet will follow the pattern described in 4.4.1.

The following example shows the ToC of three consecutive packets, each carrying three frame-blocks, in an interleaved two-channel session. Here, the two channels are left (L) and right (R) with L coming before R, and the interleaving length is 3 (i.e., ILL=2). This results in the interleaving group size of 9 frame-blocks.

Packet #1

ILL=2, ILP=0:
+-----+-----+-----+-----+
| 1L | 1R | 4L | 4R | 7L | 7R |
+-----+-----+-----+-----+
|<----->|<----->|<----->|
Frame- Frame- Frame-
Block 1 Block 4 Block 7

Packet #2

ILL=2, ILP=1:
+-----+-----+-----+-----+
| 2L | 2R | 5L | 5R | 8L | 8R |
+-----+-----+-----+-----+
|<----->|<----->|<----->|
Frame- Frame- Frame-
Block 2 Block 5 Block 8

Packet #3

ILL=2, ILP=2:
+-----+-----+-----+-----+
| 3L | 3R | 6L | 6R | 9L | 9R |
+-----+-----+-----+-----+
|<----->|<----->|<----->|
Frame- Frame- Frame-
Block 3 Block 6 Block 9

A ToC entry takes the following format in octet-aligned mode:

```

 0 1 2 3 4 5 6 7
+-----+
|F|  FT  |Q|P|P|
+-----+
```

F (1 bit): see definition in Section 4.3.2.

FT (4 bits, unsigned integer): see definition in Section 4.3.2.

Q (1 bit): see definition in Section 4.3.2.

P bits: padding bits, MUST be set to zero, and MUST be ignored on reception.

The list of CRCs is OPTIONAL. It only exists if the use of CRC is signalled out-of-band for the session. When present, each CRC in the list is 8 bits long and corresponds to a speech frame (NOT a frame-block) carried in the payload. Calculation and use of the CRC is specified in the next section.

4.4.2.1. Use of Frame CRC for UED over IP

The general concept of UED/UEP over IP is discussed in Section 3.6. This section provides more details on how to use the frame CRC in the octet-aligned payload header together with a partial transport layer checksum to achieve UED.

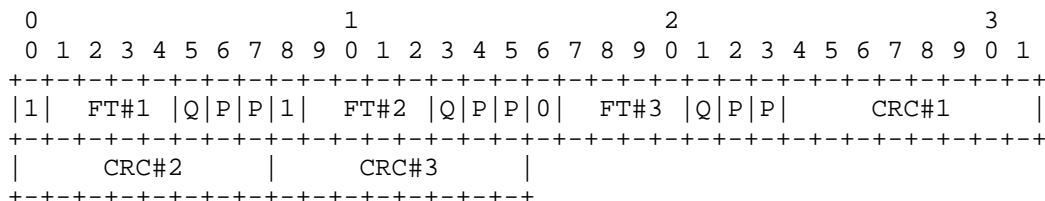
To achieve UED, one SHOULD use a transport layer checksum (for example, the one defined in UDP-Lite [19]) to protect the IP, transport protocol (e.g., UDP-Lite), and RTP headers, as well as the payload header and the table of contents in the payload. The frame CRC, when used, MUST be calculated only over all class A bits in the AMR or AMR-WB frame. Class B and C bits in the AMR or AMR-WB frame MUST NOT be included in the CRC calculation and SHOULD NOT be covered by the transport checksum.

Note, the number of class A bits for various coding modes in AMR codec is specified as informative in [2] and is therefore copied into Table 1 in Section 3.6 to make it normative for this payload format. The number of class A bits for various coding modes in AMR-WB codec is specified as normative in Table 2 in [4], and the SID frame (FT=9) has 40 class A bits. These definitions of class A bits MUST be used for this payload format.

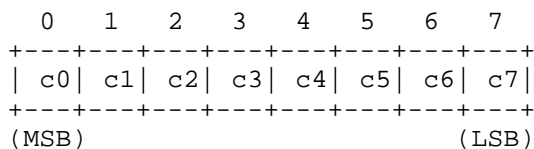
If the transport layer checksum or link layer checksum detects any errors within the protected (sensitive) part, it is assumed that the complete packet will be discarded as defined by UDP-Lite [19].

The receiver of the payload SHOULD examine the data integrity of the received class A bits by re-calculating the CRC over the received class A bits and comparing the result to the value found in the received payload header. If the two values mismatch, the receiver SHALL consider the class A bits in the receiver frame damaged and MUST clear the Q flag of the frame (i.e., set it to 0). This will subsequently cause the frame to be marked as SPEECH_BAD, if the FT of the frame is 0..7 for AMR or 0..8 for AMR-WB, or SID_BAD if the FT of the frame is 8 for AMR or 9 for AMR-WB, before it is passed to the speech decoder. See [6] and [7] more details.

The following example shows an octet-aligned ToC with a CRC list for a payload containing 3 speech frames from a single-channel session (assuming none of the FTs is equal to 14 or 15):



Each of the CRCs takes 8 bits



and is calculated by the cyclic generator polynomial,

$$C(x) = 1 + x^2 + x^3 + x^4 + x^8$$

where ^ is the exponentiation operator.

In binary form, the polynomial appears as follows: 101110001 (MSB..LSB).

The actual calculation of the CRC is made as follows: First, an 8-bit CRC register is reset to zero: 00000000. For each bit over which the CRC shall be calculated, an XOR operation is made between the rightmost (LSB) bit of the CRC register and the bit. The CRC

register is then right-shifted one step (each bit's significance is reduced by one), inputting a "0" as the leftmost bit (MSB). If the result of the XOR operation mentioned above is a "1", then "10111000" is bit-wise XOR-ed into the CRC register. This operation is repeated for each bit that the CRC should cover. In this case, the first bit would be d(0) for the speech frame for which the CRC should cover. When the last bit (e.g., d(54) for AMR 5.9 according to Table 1 in Section 3.6) has been used in this CRC calculation, the contents in CRC register should simply be copied to the corresponding field in the list of CRCs.

Fast calculation of the CRC on a general-purpose CPU is possible using a table-driven algorithm.

4.4.3. Speech Data

In octet-aligned mode, speech data is carried in a similar way to that in the bandwidth-efficient mode as discussed in Section 4.3.3, with the following exceptions:

- The last octet of each speech frame MUST be padded with zero bits at the end if all bits in the octet are not used. The padding bits MUST be ignored on reception. In other words, each speech frame MUST be octet-aligned.
- When multiple speech frames are present in the speech data (i.e., compound payload), the speech frames are arranged either one whole frame after another as usual, or with the octets of all frames interleaved together at the octet level, depending on the media type parameters negotiated for the payload type. Since the bits within each frame are ordered with the most error-sensitive bits first, interleaving the octets collects those sensitive bits from all frames to be nearer the beginning of the packet. This is called "robust sorting order" which allows the application of UED (such as UDP-Lite [19]) or UEP (such as the ULP [22]) mechanisms to the payload data. The details of assembling the payload are given in the next section.

The use of robust sorting order for a payload type MUST be agreed via out-of-band means. Section 8 specifies a media type parameter for this purpose.

Note, robust sorting order MUST only be performed on the frame level and thus is independent of interleaving, which is at the frame-block level, as described in Section 4.4.1. In other words, robust sorting can be applied to either non-interleaved or interleaved payload types.

4.4.4. Methods for Forming the Payload

Two different packetization methods, namely, normal order and robust sorting order, exist for forming a payload in octet-aligned mode. In both cases, the payload header and table of contents are packed into the payload the same way; the difference is in the packing of the speech frames.

The payload begins with the payload header of one octet, or two octets if frame interleaving is selected. The payload header is followed by the table of contents consisting of a list of one-octet ToC entries. If frame CRCs are to be included, they follow the table of contents with one 8-bit CRC filling each octet. Note that if a given frame has a ToC entry with FT=14 or 15, there will be no CRC present.

The speech data follows the table of contents, or the CRCs if present. For packetization in the normal order, all of the octets comprising a speech frame are appended to the payload as a unit. The speech frames are packed in the same order as their corresponding ToC entries are arranged in the ToC list, with the exception that if a given frame has a ToC entry with FT=14 or 15, there will be no data octets present for that frame.

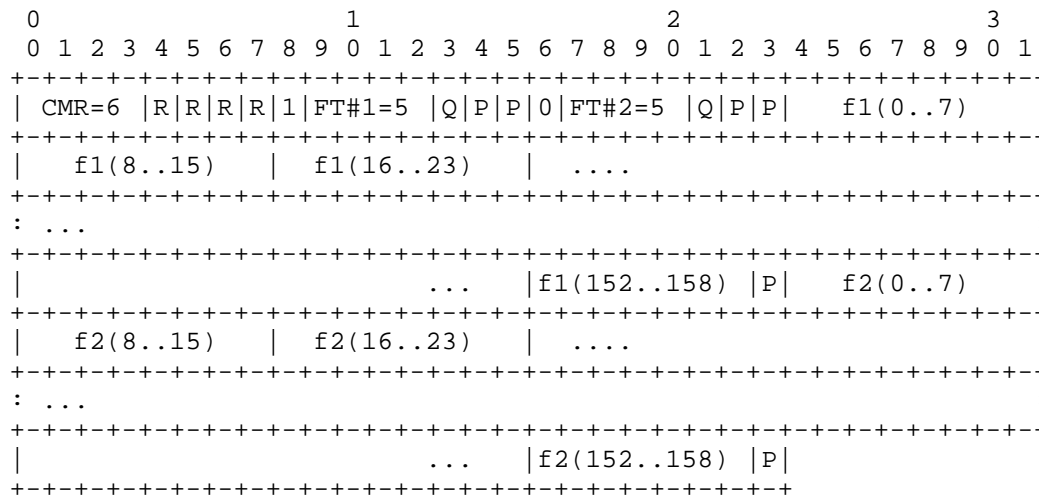
For packetization in robust sorting order, the octets of all speech frames are interleaved together at the octet level. That is, the data portion of the payload begins with the first octet of the first frame, followed by the first octet of the second frame, then the first octet of the third frame, and so on. After the first octet of the last frame has been appended, the cycle repeats with the second octet of each frame. The process continues for as many octets as are present in the longest frame. If the frames are not all the same octet length, a shorter frame is skipped once all octets in it have been appended. The order of the frames in the cycle will be sequential if frame interleaving is not in use, or according to the interleave pattern specified in the payload header if frame interleaving is in use. Note that if a given frame has a ToC entry with FT=14 or 15, there will be no data octets present for that frame, so it is skipped in the robust sorting cycle.

The UED and/or UEP is RECOMMENDED to cover at least the RTP header, payload header, table of contents, and class A bits of a sorted payload. Exactly how many octets need to be covered depends on the network and application. If CRCs are used together with robust sorting, only the RTP header, the payload header, and the ToC SHOULD be covered by UED/UEP. The means for communicating the number of octets to be covered to other layers performing UED/UEP is beyond the scope of this specification.

4.4.5. Payload Examples

4.4.5.1. Basic Single-Channel Payload Carrying Multiple Frames

The following diagram shows an octet aligned payload from a single channel payload type that carries two AMR frames of 7.95 kbps coding mode (FT=5). In the payload, a codec mode request is sent (CMR=6), requesting the encoder at the receiver's side to use AMR 10.2 kbps coding mode. No frame CRC, interleaving, or robust sorting is in use.



Note, in the above example, the last octet in both speech frames is padded with one zero bit to make it octet-aligned.

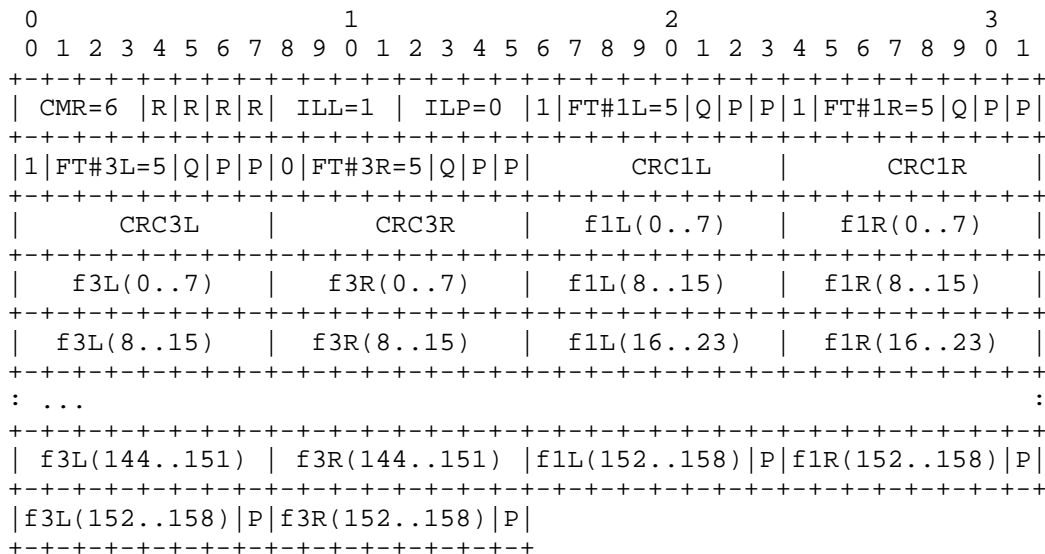
4.4.5.2. Two-Channel Payload with CRC, Interleaving, and Robust Sorting

This example shows an octet aligned payload from a two-channel payload type. Two frame-blocks, each containing two speech frames of 7.95 kbps coding mode (FT=5), are carried in this payload.

The two channels are left (L) and right (R) with L coming before R. In the payload, a codec mode request is also sent (CMR=6), requesting the encoder at the receiver's side to use AMR 10.2 kbps coding mode.

Moreover, frame CRC, robust sorting, and frame-block interleaving are all enabled for the payload type. The interleaving length is 2 (ILL=1), and this payload is the first one in an interleaving group (ILP=0).

The first two frames in the payload are the L and R channel speech frames of frame-block #1, consisting of bits f1L(0..158) and f1R(0..158), respectively. The next two frames are the L and R channel frames of frame-block #3, consisting of bits f3L(0..158) and f3R(0..158), respectively, due to interleaving. For each of the four speech frames, a CRC is calculated as CRC1L(0..7), CRC1R(0..7), CRC3L(0..7), and CRC3R(0..7), respectively. Finally, the payload is robust sorted.



Note, in the above example, the last octet in all four speech frames is padded with one zero bit to make it octet-aligned.

4.5. Implementation Considerations

An application implementing this payload format MUST understand all the payload parameters in the out-of-band signaling used. For example, if an application uses SDP, all the SDP and media type parameters in this document MUST be understood. This requirement ensures that an implementation always can decide if it is capable or not of communicating.

No operating mode of the payload format is mandatory to implement. The requirements of the application using the payload format should be used to determine what to implement. To achieve basic interoperability, an implementation SHOULD at least implement both bandwidth-efficient and octet-aligned modes for a single audio

channel. The other operating modes: interleaving, robust sorting, and frame-wise CRC (in both single and multi-channel) are OPTIONAL to implement.

The mode-change-period, mode-change-capability, and mode-change-neighbor parameters are intended for signaling with GSM endpoints. When interoperability with GSM is desired, encoders SHOULD only perform codec mode changes to neighboring modes and in integer multiples of 40 ms (two frame-blocks), but decoders SHOULD accept codec mode changes at any time, i.e., for every frame-block. The encoder may arbitrarily select the initial phase (odd or even frame-block) where codec mode changes are performed, but then SHOULD stick to that phase as far as possible. However, in rare cases, handovers or other events (e.g., call forwarding) may change this phase and may also cause mode changes to non-neighboring modes. The decoder SHALL therefore be prepared to accept changes also in the other phase and to other modes. Section 8 specifies the usage of the parameters mode-change-period and mode-change-capability to indicate the desired behavior in applications.

See 3GPP TS 26.103 [28] for preferred AMR and AMR-WB configurations for operation in GSM and 3GPP UMTS networks. In gateway scenarios, encoders can be requested through the "mode-set" parameter to use a limited mode-set that is supported by the link beyond the gateway. Further, to avoid congestion on that link, the encoder SHOULD limit the initial codec mode for a session to a lower mode, until at least one frame-block is received with rate control information.

4.5.1. Decoding Validation

When processing a received payload packet, if the receiver finds that the calculated payload length, based on the information for the payload type and the values found in the payload header fields, does not match the size of the received packet, the receiver SHOULD discard the packet. This is because decoding a packet that has errors in its length field could severely degrade the speech quality.

5. AMR and AMR-WB Storage Format

The storage format is used for storing AMR or AMR-WB speech frames in a file or as an email attachment. Multiple channel content is supported.

In general, an AMR or AMR-WB file has the following structure:

```
+-----+
| Header |
+-----+
| Speech frame 1 |
+-----+
: ... :
+-----+
| Speech frame n |
+-----+
```

Note, to preserve interoperability with already deployed implementations, single-channel content uses a file header format different from that of multi-channel content.

There also exists another storage format for AMR and AMR-WB that is suitable for applications with more advanced demands on the storage format, like random access or synchronization with video. This format is the 3GPP-specified ISO-based multimedia file format 3GP [31]. Its media type is specified by RFC 3839 [32].

5.1. Single-Channel Header

A single-channel AMR or AMR-WB file header contains only a magic number. Different magic numbers are defined to distinguish AMR from AMR-WB.

The magic number for single-channel AMR files MUST consist of ASCII character string:

```
"#!AMR\n"
(or 0x2321414d520a in hexadecimal).
```

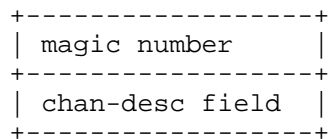
The magic number for single-channel AMR-WB files MUST consist of ASCII character string:

```
"#!AMR-WB\n"
(or 0x2321414d522d57420a in hexadecimal).
```

Note, the "\n" is an important part of the magic numbers and MUST be included in the comparison, since, otherwise, the single-channel magic numbers above will become indistinguishable from those of the multi-channel files defined in the next section.

5.2. Multi-Channel Header

The multi-channel header consists of a magic number followed by a 32-bit channel description field, giving the multi-channel header the following structure:



The magic number for multi-channel AMR files MUST consist of the ASCII character string:

```

"#!AMR_MCl.0\n"
(or 0x2321414d525F4D43312E300a in hexadecimal).

```

The magic number for multi-channel AMR-WB files MUST consist of the ASCII character string:

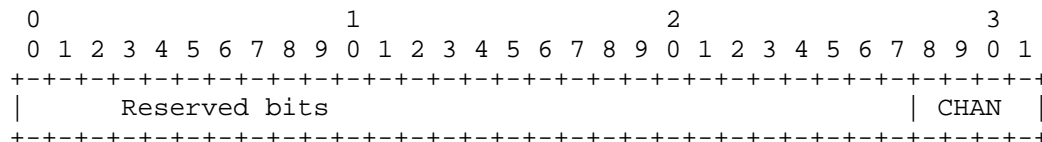
```

"#!AMR-WB_MCl.0\n"
(or 0x2321414d522d57425F4D43312E300a in hexadecimal).

```

The version number in the magic numbers refers to the version of the file format.

The 32 bit channel description field is defined as:



Reserved bits: MUST be set to 0 when written, and a reader MUST ignore them.

CHAN (4 bits, unsigned integer): Indicates the number of audio channels contained in this storage file. The valid values and the order of the channels within a frame-block are specified in Section 4.1 in [12].

6. Congestion Control

The general congestion control considerations for transporting RTP data apply to AMR or AMR-WB speech over RTP as well. However, the multi-rate capability of AMR and AMR-WB speech coding may provide an advantage over other payload formats for controlling congestion since the bandwidth demand can be adjusted by selecting a different coding mode.

Another parameter that may impact the bandwidth demand for AMR and AMR-WB is the number of frame-blocks that are encapsulated in each RTP payload. Packing more frame-blocks in each RTP payload can reduce the number of packets sent and hence the overhead from IP/UDP/RTP headers, at the expense of increased delay.

If forward error correction (FEC) is used to combat packet loss, the amount of redundancy added by FEC will need to be regulated so that the use of FEC itself does not cause a congestion problem.

It is RECOMMENDED that AMR or AMR-WB applications using this payload format employ congestion control. The actual mechanism for congestion control is not specified but should be suitable for real-time flows, possibly "TCP Friendly Rate Control" [21].

7. Security Considerations

RTP packets using the payload format defined in this specification are subject to the general security considerations discussed in [8] and in any used profile, like AVP [12] or SAVP [26].

As this format transports encoded speech, the main security issues include confidentiality, authentication, and integrity of the speech itself. The payload format itself does not have any built-in security mechanisms. External mechanisms, such as SRTP [26], need to be used for this functionality. Note that the appropriate mechanism to provide security to RTP and the payloads following this memo may vary. It is dependent on the application, the transport, and the signaling protocol employed. Therefore, a single mechanism is not sufficient, although if suitable the usage of SRTP [26] is RECOMMENDED. Other known mechanisms that may be used are IPsec [33] and TLS [34] (RTP over TCP), but other alternatives may also exist.

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

7.1. Confidentiality

To achieve confidentiality of the encoded AMR or AMR-WB speech, all speech data bits will need to be encrypted. There is less of a need to encrypt the payload header or the table of contents due to a) that they only carry information about the requested speech mode, frame type, and frame quality, and b) that this information could be useful to some third party, e.g., quality monitoring.

The packetization and unpacketization of the AMR and AMR-WB payload is done only at the endpoints. Therefore encryption should be performed after packet encapsulation, and decryption should be performed before packet decapsulation.

Encryption may affect interleaving. Specifically, a change of keys should occur at the boundary between interleaving groups. If it is not done at that boundary on both endpoints, the speech quality will be degraded during the complete interleaving group for any receiver.

The encryption mechanism may impact the robustness of the error correcting mechanism. This is discussed in Section 9.5 of SRTP [26]. From this, UED/UEP based on robust sorting may be difficult to apply when the payload data is encrypted.

7.2. Authentication and Integrity

To authenticate the sender and to protect the integrity of the RTP packets in transit, an external mechanism has to be used. As stated before, it is RECOMMENDED that SRTP [26] be used for common interoperability. Note that the use of UED/UEP may be difficult to combine with some integrity protection mechanisms because any bit errors will cause the integrity check to fail.

Data tampering by a man-in-the-middle attacker could result in erroneous depacketization/decoding that could lower the speech quality or produce unintelligible communications. Tampering with the CMR field may result in a different speech quality than desired.

8. Payload Format Parameters

This section defines the parameters that may be used to select optional features of the AMR and AMR-WB payload formats. The parameters are defined here as part of the media type registrations for the AMR and AMR-WB speech codecs. The registrations are done following RFC 4855 [15] and the media registration rules [14].

A mapping of the parameters into the Session Description Protocol (SDP) [11] is also provided for those applications that use SDP. Equivalent parameters could be defined elsewhere for use with control protocols that do not use media types or SDP.

Two separate media type registrations are made, one for AMR and one for AMR-WB, because they are distinct encodings that must be distinguished by their own media type.

Data formats are specified for both real-time transport in RTP and for storage type applications such as email attachments.

8.1. AMR Media Type Registration

The media type for the Adaptive Multi-Rate (AMR) codec is allocated from the IETF tree since AMR is a widely used speech codec in general VoIP and messaging applications. This media type registration covers both real-time transfer via RTP and non-real-time transfers via stored files.

Note, any unspecified parameter MUST be ignored by the receiver.

Media Type name: audio

Media subtype name: AMR

Required parameters: none

Optional parameters:

These parameters apply to RTP transfer only.

octet-align: Permissible values are 0 and 1. If 1, octet-aligned operation SHALL be used. If 0 or if not present, bandwidth-efficient operation is employed.

mode-set: Restricts the active codec mode set to a subset of all modes, for example, to be able to support transport channels such as GSM networks in gateway use cases. Possible values are a comma separated list of modes from the set: 0,...,7 (see Table 1a [2]). The SID frame type 8 and NO_DATA (frame type 15) are never included in the mode set, but can always be used. If mode-set is specified, it MUST be abided, and frames encoded with modes outside of the subset MUST NOT be sent in any RTP payload or used in codec mode requests. If not present, all codec modes are allowed for the payload type.

mode-change-period: Specifies a number of frame-blocks, N (1 or 2), that is the frame-block period at which codec mode changes are allowed for the sender. The initial phase of the interval is arbitrary, but changes must be separated by a period of N frame-blocks, i.e., a value of 2 allows the sender to change mode every second frame-block. The value of N SHALL be either 1 or 2. If this parameter is not present, mode changes are allowed at any time during the session, i.e., N=1.

mode-change-capability: Specifies if the client is capable to transmit with a restricted mode change period. The parameter may take value of 1 or 2. A value of 1 indicates that the client is not capable of restricting the mode change period to 2, and that the codec mode may be changed at any point. A value of 2 indicates that the client has the capability to restrict the mode change period to 2, and thus that the client can correctly interoperate with a receiver requiring a mode-change-period=2. If this parameter is not present, the mode-change restriction capability is not supported, i.e. mode-change-capability=1. To be able to interoperate fully with gateways to circuit switched networks (for example, GSM networks), transmissions with restricted mode changes (mode-change-capability=2) are required. Thus, clients RECOMMENDED to have the capability to support transmission according to mode-change-capability=2.

mode-change-neighbor: Permissible values are 0 and 1. If 1, the sender SHOULD only perform mode changes to the neighboring modes in the active codec mode set.

Neighboring modes are the ones closest in bit rate to the current mode, either the next higher or next lower rate. If 0 or if not present, change between any two modes in the active codec mode set is allowed.

maxptime: The maximum amount of media which can be encapsulated in a payload packet, expressed as time in milliseconds. The time is calculated as the sum of the time that the media present in the packet represents. The time SHOULD be an integer multiple of the frame size. If this parameter is not present, the sender MAY encapsulate any number of speech frames into one RTP packet.

crc: Permissible values are 0 and 1. If 1, frame CRCs SHALL be included in the payload. If 0 or not present, CRCs SHALL NOT be used. If crc=1, this also implies automatically that octet-aligned operation SHALL be used for the session.

robust-sorting: Permissible values are 0 and 1. If 1, the payload SHALL employ robust payload sorting. If 0 or if not present, simple payload sorting SHALL be used. If robust-sorting=1, this also implies automatically that octet-aligned operation SHALL be used for the session.

interleaving: Indicates that frame-block level interleaving SHALL be used for the session, and its value defines the maximum number of frame-blocks allowed in an interleaving group (see Section 4.4.1). If this parameter is not present, interleaving SHALL NOT be used. The presence of this parameter also implies automatically that octet-aligned operation SHALL be used.

ptime: see RFC 4566 [11].

channels: The number of audio channels. The possible values (1-6) and their respective channel order is specified in Section 4.1 in [12]. If omitted, it has the default value of 1.

max-red: The maximum duration in milliseconds that elapses between the primary (first) transmission of a frame and any redundant transmission that the sender will use. This parameter allows a receiver to have a bounded delay when redundancy is used. Allowed values are between 0 (no redundancy will be used) and 65535. If the parameter is omitted, no limitation on the use of redundancy is present.

Encoding considerations:

The Audio data is binary data, and must be encoded for non-binary transport; the Base64 encoding is suitable for email. When used in RTP context the data is framed as defined in [14].

Security considerations:

See Section 7 of RFC 4867.

Public specification:

RFC 4867
3GPP TS 26.090, 26.092, 26.093, 26.101

Applications that use this media type:

This media type is used in numerous applications needing transport or storage of encoded voice. Some examples include; Voice over IP, streaming media, voice messaging, and voice recording on digital cameras.

Additional information:

The following applies to stored-file transfer methods:

Magic numbers:

single-channel:

ASCII character string "#!AMR\n"
(or 0x2321414d520a in hexadecimal)

multi-channel:

ASCII character string "#!AMR_MCl.0\n"
(or 0x2321414d525F4D43312E300a in hexadecimal)

File extensions: amr, AMR

Macintosh file type code: "amr " (fourth character is space)

AMR speech frames may also be stored in the file format "3GP" defined in 3GPP TS 26.244 [31], which is identified using the media types "audio/3GPP" or "video/3GPP" as registered by RFC 3839 [32].

Person & email address to contact for further information:

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Ari Lakaniemi <ari.lakaniemi@nokia.com>

Intended usage: COMMON.

This media type is widely used in streaming, VoIP, and messaging applications on many types of devices.

Restrictions on usage:

When this media type is used in the context of transfer over RTP, the RTP payload format specified in Section 4 SHALL be used. In all other contexts, the file format defined in Section 5 SHALL be used.

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Change controller:

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

8.2. AMR-WB Media Type Registration

The media type for the Adaptive Multi-Rate Wideband (AMR-WB) codec is allocated from the IETF tree since AMR-WB is a widely used speech codec in general VoIP and messaging applications. This media type registration covers both real-time transfer via RTP and non-real-time transfers via stored files.

Note, any unspecified parameter MUST be ignored by the receiver.

Media Type name: audio

Media subtype name: AMR-WB

Required parameters: none

Optional parameters:

These parameters apply to RTP transfer only.

octet-align: Permissible values are 0 and 1. If 1, octet-aligned operation SHALL be used. If 0 or if not present, bandwidth-efficient operation is employed.

mode-set: Restricts the active codec mode set to a subset of all modes, for example, to be able to support transport channels such as GSM networks in gateway use cases. Possible values are a comma-separated list of modes from the set: 0,...,8 (see Table 1a [4]). The SID frame type 9, SPEECH_LOST (frame type 14), and NO_DATA (frame type 15) are never included in the mode set, but can always be used. If mode-set is specified, it MUST be abided, and frames encoded with modes outside of the subset MUST NOT be sent in any RTP payload or used in codec mode requests. If not present, all codec modes are allowed for the payload type.

mode-change-period: Specifies a number of frame-blocks, N (1 or 2), that is the frame-block period at which codec mode changes are allowed for the sender. The initial phase of the interval is arbitrary, but changes must be separated by multiples of N frame-blocks, i.e., a value of 2 allows the sender to change mode every second frame-block. The value of N SHALL be either 1 or 2. If this parameter is not present, mode changes are allowed at Any time during the session, i.e., N=1.

mode-change-capability: Specifies if the client is capable to transmit with a restricted mode change period. The parameter may take value of 1 or 2. A value of 1 indicates that the client is not capable of restricting the mode change period to 2, and that the codec mode may be changed at any point. A value of 2 indicates that the client has the capability to restrict the mode change period to 2, and thus that the client can correctly interoperate with a receiver requiring a mode-change-period=2. If this parameter is not present, the mode-change restriction capability is not supported, i.e. mode-change-capability=1. To be able to interoperate fully with gateways to circuit switched networks (for example, GSM networks), transmissions with restricted mode changes (mode-change-capability=2) are required. Thus, clients are RECOMMENDED to have the capability to support transmission according to mode-change-capability=2.

mode-change-neighbor: Permissible values are 0 and 1. If 1, the sender SHOULD only perform mode changes to the neighboring modes in the active codec mode set. Neighboring modes are the ones closest in bit rate to the current mode, either the next higher or next lower rate. If 0 or if not present, change between any two modes in the active codec mode set is allowed.

maxptime: The maximum amount of media which can be encapsulated in a payload packet, expressed as time in milliseconds. The time is calculated as the sum of the time that the media present in the packet represents. The time SHOULD be an integer multiple of the frame size. If this parameter is not present, the sender MAY encapsulate any number of speech frames into one RTP packet.

crc: Permissible values are 0 and 1. If 1, frame CRCs SHALL be included in the payload. If 0 or not present, CRCs SHALL NOT be used. If crc=1, this also implies automatically that octet-aligned operation SHALL be used for the session.

robust-sorting: Permissible values are 0 and 1. If 1, the payload SHALL employ robust payload sorting. If 0 or if not present, simple payload sorting SHALL be used. If robust-sorting=1, this also implies automatically that octet-aligned operation SHALL be used for the session.

interleaving: Indicates that frame-block level interleaving SHALL be used for the session, and its value defines the maximum number of frame-blocks allowed in an interleaving group (see Section 4.4.1). If this parameter is not present, interleaving SHALL NOT be used. The presence of this parameter also implies automatically that octet-aligned operation SHALL be used.

ptime: see RFC 2327 [11].

channels: The number of audio channels. The possible values (1-6) and their respective channel order is specified in Section 4.1 in [12]. If omitted, it has the default value of 1.

max-red: The maximum duration in milliseconds that elapses between the primary (first) transmission of a frame and any redundant transmission that the sender will use. This parameter allows a receiver to have a bounded delay when redundancy is used. Allowed values are between 0 (no redundancy will be used) and 65535. If the parameter is omitted, no limitation on the use of redundancy is present.

Encoding considerations:

The Audio data is binary data, and must be encoded for non-binary transport; the Base64 encoding is suitable for email. When used in RTP context the data is framed as defined in [14].

Security considerations:

See Section 7 of RFC 4867.

Public specification:

RFC 4867
3GPP TS 26.190, 26.192, 26.193, 26.201

Applications that use this media type:

This media type is used in numerous applications needing transport or storage of encoded voice. Some examples include; Voice over IP, streaming media, voice messaging, and voice recording on digital cameras.

Additional information:

The following applies to stored-file transfer methods:

Magic numbers:

single-channel:

ASCII character string "#!AMR-WB\n"

(or 0x2321414d522d57420a in hexadecimal)

multi-channel:

ASCII character string "#!AMR-WB_MCl.0\n"

(or 0x2321414d522d57425F4D43312E300a in hexadecimal)

File extensions: awb, AWB

Macintosh file type code: amrw

Object identifier or OID: none

AMR-WB speech frames may also be stored in the file format "3GP" defined in 3GPP TS 26.244 [31] and identified using the media type "audio/3GPP" or "video/3GPP" as registered by RFC 3839 [32].

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Ari Lakaniemi <ari.lakaniemi@nokia.com>

Intended usage: COMMON.

This media type is widely used in streaming, VoIP, and messaging applications on many types of devices.

Restrictions on usage:

When this media type is used in the context of transfer over RTP, the RTP payload format specified in Section 4 SHALL be used. In all other contexts, the file format defined in Section 5 SHALL be used.

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Change controller:

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

8.3. 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) [11], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the AMR or AMR-WB codec, the mapping is as follows:

- The media type ("audio") goes in SDP "m=" as the media name.
- The media subtype (payload format name) goes in SDP "a=rtpmap" as the encoding name. The RTP clock rate in "a=rtpmap" MUST be 8000 for AMR and 16000 for AMR-WB, and the encoding parameters (number of channels) MUST either be explicitly set to N or omitted, implying a default value of 1. The values of N that are allowed are specified in Section 4.1 in [12].
- The parameters "ptime" and "maxptime" go in the SDP "a=ptime" and "a=maxptime" attributes, respectively.
- Any remaining parameters go in the SDP "a=fmtp" attribute by copying them directly from the media type parameter string as a semicolon-separated list of parameter=value pairs.

8.3.1. Offer-Answer Model Considerations

The following considerations apply when using SDP Offer-Answer procedures to negotiate the use of AMR or AMR-WB payload in RTP:

- Each combination of the RTP payload transport format configuration parameters (octet-align, crc, robust-sorting, interleaving, and channels) is unique in its bit-pattern and not compatible with any other combination. When creating an offer in an application desiring to use the more advanced features (crc, robust-sorting, interleaving, or more than one channel), the offerer is RECOMMENDED to also offer a payload type containing only the octet-aligned or bandwidth-efficient configuration with a single channel. If multiple configurations are of interest to the application, they may all be offered; however, care should be taken not to offer too many payload types. An SDP answerer MUST include, in the SDP answer for a payload type, the following parameters unmodified from the SDP offer (unless it removes the payload type): "octet-align"; "crc"; "robust-sorting"; "interleaving"; and "channels". The SDP offerer and answerer MUST generate AMR or AMR-WB packets as described by these parameters.
- The "mode-set" parameter can be used to restrict the set of active AMR/AMR-WB modes used in a session. This functionality is primarily intended for gateways to access networks such as GSM or 3GPP UMTS, where the access network may be capable of supporting only a subset of AMR/AMR-WB modes. The 3GPP preferred codec configurations are defined in 3GPP TS 26.103 [25], and it is RECOMMENDED that other networks also needing to restrict the mode set follow the preferred codec configurations defined in 3GPP for greatest interoperability.

The parameter is bi-directional, i.e., the restricted set applies to media both to be received and sent by the declaring entity. If a mode set was supplied in the offer, the answerer SHALL return the mode-set unmodified or reject the payload type. However, the answerer is free to choose a mode-set in the answer only if no mode-set was supplied in the offer for a unicast two-peer session. The mode-set in the answer is binding both for offerer and answerer. Thus, an offerer supporting all modes and subsets SHOULD NOT include the mode-set parameter. For any other offerer it is RECOMMENDED to include each mode-set it can support as a separate payload type within the offer. For multicast sessions, the answerer SHALL only participate in the session if it supports the offered mode-set. Thus, it is RECOMMENDED that any offer for a multicast session include only the mode-set it will require the answerers to support, and that the mode-set be likely to be supported by all participants.

- The parameters "mode-change-period" and "mode-change-capability" are intended to be used in sessions with gateways, for example, when interoperating with GSM networks. Both parameters are declarative and are combined to allow a session participant to determine if the payload type can be supported. The mode-change-period will indicate what the offerer or answerer requires of data it receives, while the mode-change-capability indicates its transmission capabilities.

A mode-change-period=2 in the offer indicates a requirement on the answerer to send with a mode-change period of 2, i.e., support mode-change-capability=2. If the answerer requires mode-change-period=2, it SHALL only include it in the answer if the offerer either has indicated support with mode-change-capability=2 or has indicated mode-change-period=2; otherwise, the payload type SHALL be rejected. An offerer that supports mode-change-capability=2 SHALL include the parameter in all offers to ensure the greatest possible interoperability, unless it includes mode-change-period=2 in the offer. The mode-change-capability SHOULD be included in answers. It is then indicating the answerer's capability to transmit with that mode-change-period for the provided payload format configuration. The information is useful in future re-negotiation of the payload formats.

- The parameter "mode-change-neighbor" is a recommendation to restrict the switching of codec modes to its neighbor and SHOULD be followed. It is intended to be used in gateway scenarios (for example, to GSM networks) where the support of

this parameter and the operations it implies improves interoperability.

"mode-change-neighbor" is a declarative parameter. By including the parameter, the offerer or answerer indicates that it desires to receive streams with "mode-change-neighbor" restrictions.

- In most cases, the parameters "maxptime" and "ptime" will not affect interoperability; however, the setting of the parameters can affect the performance of the application. The SDP offer-answer handling of the "ptime" parameter is described in RFC 3264 [13]. The "maxptime" parameter MUST be handled in the same way.
- The parameter "max-red" is a stream property parameter. For send-only or send-recv unicast media streams, the parameter declares the limitation on redundancy that the stream sender will use. For recvonly streams, it indicates the desired value for the stream sent to the receiver. The answerer MAY change the value, but is RECOMMENDED to use the same limitation as the offer declares. In the case of multicast, the offerer MAY declare a limitation; this SHALL be answered using the same value. A media sender using this payload format is RECOMMENDED to always include the "max-red" parameter. This information is likely to simplify the media stream handling in the receiver. This is especially true if no redundancy will be used, in which case "max-red" is set to 0. As this parameter was not defined originally, some senders will not declare this parameter even if it will limit or not send redundancy at all.
- Any unknown parameter in an offer SHALL be removed in the answer.

8.3.2. Usage of Declarative SDP

In declarative usage, like SDP in RTSP [29] or SAP [30], the parameters SHALL be interpreted as follows:

- The payload format configuration parameters (octet-align, crc, robust-sorting, interleaving, and channels) are all declarative, and a participant MUST use the configuration(s) that is provided for the session. More than one configuration may be provided if necessary by declaring multiple RTP payload types; however, the number of types should be kept small.

- Any restriction of the AMR or AMR-WB encoder mode-switching and mode usage through the "mode-set", and "mode-change-period" MUST be followed by all participants of the session. The restriction indicated by "mode-change-neighbor" SHOULD be followed. Please note that such restrictions may be necessary if gateways to other transport systems like GSM participate in the session. Failure to consider such restrictions may result in failure for a peer behind such a gateway to correctly receive all or parts of the session. Also, if different restrictions are needed by different peers in the same session (unless a common subset of the restrictions exists), some peer will not be able to participate. Note that the usage of mode-change-capability is meaningless when no negotiation exists, and can thus be excluded in any declarations.
- Any "maxptime" and "ptime" values should be selected with care to ensure that the session's participants can achieve reasonable performance.
- The usage of "max-red" puts a global upper limit on the usage of redundancy that needs to be followed by all that understand the parameter. However, due to the late addition of this parameter, it may be ignored by some implementations.

8.3.3. Examples

Some example SDP session descriptions utilizing AMR and AMR-WB encodings follow. In these examples, long a=fmtp lines are folded to meet the column width constraints of this document; the backslash ("\") at the end of a line and the carriage return that follows it should be ignored.

In an example of the usage of AMR in a possible GSM gateway-to-gateway scenario, the offerer is capable of supporting three different mode-sets and needs the mode-change-period to be 2 in combination with mode-change-neighbor restrictions. The other gateway can only support two of these mode-sets and removes the payload type 97 in the answer. If the offering GSM gateway only supports a single mode-set active at the same time, it should consider doing the 1 out of N selection procedures described in Section 10.2 of [13]:

Offer:

```
m=audio 49120 RTP/AVP 97 98 99
a=rtpmap:97 AMR/8000/1
a=fmtp:97 mode-set=0,2,5,7; mode-change-period=2; \
  mode-change-capability=2; mode-change-neighbor=1
a=rtpmap:98 AMR/8000/1
a=fmtp:98 mode-set=0,2,3,6; mode-change-period=2; \
  mode-change-capability=2; mode-change-neighbor=1
a=rtpmap:99 AMR/8000/1
a=fmtp:99 mode-set=0,2,3,4; mode-change-period=2; \
  mode-change-capability=2; mode-change-neighbor=1
a=maxptime:20
```

Answer:

```
m=audio 49120 RTP/AVP 98 99
a=rtpmap:98 AMR/8000/1
a=fmtp:98 mode-set=0,2,3,6; mode-change-period=2; \<
  mode-change-capability=2; mode-change-neighbor=1
a=rtpmap:99 AMR/8000/1
a=fmtp:99 mode-set=0,2,3,4; mode-change-period=2; \
  mode-change-capability=2; mode-change-neighbor=1
a=maxptime:20
```

The following example shows the usage of AMR between a non-GSM endpoint and a GSM gateway. The non-GSM offerer requires no restrictions of the mode-change-period or mode-change-neighbor, but must signal its mode-change-capability in the offer and abide by those restrictions in the answer.

Offer:

```
m=audio 49120 RTP/AVP 97
a=rtpmap:97 AMR/8000/1
a=fmtp:97 mode-change-capability=2
a=maxptime:20
```

Answer:

```
m=audio 49120 RTP/AVP 97
a=rtpmap:97 AMR/8000/1
a=fmtp:97 mode-set=0,2,4,7; mode-change-period=2; \
  mode-change-capability=2; mode-change-neighbor=1
a=maxptime:20
```

Example of usage of AMR-WB in a possible VoIP scenario where UEP may be used (99) and a fallback declaration (98):

```
m=audio 49120 RTP/AVP 99 98
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 octet-align=1; mode-change-capability=2
a=rtpmap:99 AMR-WB/16000
a=fmtp:99 octet-align=1; crc=1; mode-change-capability=2
```

Example of usage of AMR-WB in a possible streaming scenario (two channel stereo):

```
m=audio 49120 RTP/AVP 99
a=rtpmap:99 AMR-WB/16000/2
a=fmtp:99 interleaving=30
a=maxptime:100
```

Note that the payload format (encoding) names are commonly shown in upper case. MIME subtypes are commonly shown in lower case. 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.

9. IANA Considerations

Two media types (audio/AMR and audio/AMR-WB) have been updated; see Section 8.

10. Changes from RFC 3267

The differences between RFC 3267 and this document are as follows:

- Added clarification of behavior in regards to mode change period and mode-change neighbor that is expected from an IP client; see Section 4.5.
- Updated the maxptime for better clarification. The sentence that previously read: "The time SHOULD be a multiple of the frame size." now says "The time SHOULD be an integer multiple of the frame size." This should have no impact on interoperability.
- Updated the definition of the mode-set parameter for clarification.
- Restricted the values for mode-change-period to 1 or 2, which are the values used in circuit-switched AMR systems.

- Added a new media type parameter Mode-Change-Capability that defaults to 1, which is the assumed behavior of any non-updated implementation. This enables the offer-answer procedures to work.
- Changed mode-change-neighbor to indicate a recommended behavior rather than a required one.
- Added an Offer-Answer Section, see Section 8.3.1. This will have implications on the interoperability to implementations that have guessed how to perform offer/answer negotiation of the payload parameters.
- Clarified and aligned the unequal detection usage with the published UDP-Lite specification in Sections 3.6.1 and 4.4.2.1. This included replacing a normative statement about packet handling with an informative paragraph with a reference to UDP-Lite.
- Clarified the bit order in the CRC calculation in Section 4.4.2.1.
- Corrected the reference in Section 5.3 for the Q and FT fields.
- Changed the padding bit definition in Sections 4.4.2 and 5.3 so that it is clear that they shall be ignored.
- Added a clarification that comfort noise frames with frame type 9, 10, and 11 SHALL NOT be used in the AMR file format.
- Clarified in Section 4.3.2 that the rules about not sending NO_DATA frames do apply for all payload format configurations with the exception of the interleaved mode.
- The reference list has been updated to now published RFCs: RFC 3448, RFC 3550, RFC 3551, RFC 3711, RFC 3828, and RFC 4566. A reference to 3GPP TS 26.101 has also been added.
- Added notes in storage format section and media type registration that AMR and AMR-WB frames can also be stored in the 3GP file format.
- Added a media type parameter "max-red" that allows the sender to declare a bounded usage of redundancy. This parameter allows a receiver to optimize its function as it will know if redundancy will be used or not. If it is used, the maximum extra delay introduced by the sender (that is needed to be considered by the receiver to fully utilize the redundancy) will be known. The addition of this parameter should have no negative effects on older implementations as they are mandated to ignore unknown

parameters per RFC 3267. In addition, older implementations are required to operate as if the value of max-red is unknown and possibly infinite.

- Updated the media type registration to comply with the new registration rules.
- Moved section on decoding validation from Security Considerations to Implementation Considerations, where it makes more sense.
- Clarified the application of encryption, integrity protection, and authentication mechanism to the payload.

11. Acknowledgements

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