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Comments are solicited and should be addressed to the author and/or the IETF Audio/Video Transport working group’s mailing list at rem-conf@es.net.

Abstract

It is required to demonstrate interoperability of implementations in order to move the RTP audio/video profile to draft standard. This memo outlines those features to be tested, as the first stage of an interoperability statement.

1 Introduction

The Internet standards process [1] places a number of requirements on a standards track protocol specification. In particular, when advancing a protocol from proposed standard to draft standard it is necessary to demonstrate at least two independent and interoperable implementations, from different code bases, of all options and features of that protocol. Further, in cases where one or more options or features have not been demonstrated in at least two interoperable
implementations, the specification may advance to the draft standard level only if those options or features are removed. The RTP Profile for Audio and Video Conferences with Minimal Control was originally specified in RFC1890 as a proposed standard [2]. The revision of this specification for draft standard status is now well underway, so it has become necessary to conduct such an interoperability demonstration.

This memo describes the set of features and options of the RTP Audio/Video profile which need to be tested as a basis for this demonstration.

This memo is for information only and does not specify a standard of any kind.

2 Features and options required to demonstrate interoperability

In order to demonstrate interoperability it is required to produce a statement of interoperability for each feature noted below. Such a statement should note the pair of implementations tested, including version numbers, and a pass/fail statement for each feature. It is not expected that every implementation will implement every feature, but each feature needs to be demonstrated by some pair of applications.

The RTP specification [3] enumerates a number of items that can be specified or modified by a profile. Those modified from the defaults by the audio/video profile are as follows, and interoperable behaviour must be demonstrated:

1. Exchange of RTCP packets with the default RTCP reporting interval.

2. Exchange of RTCP packets with a modified RTCP reporting interval as selected by a different fraction of the session bandwidth (the means by which this interval is signalled are outside of the scope of this memo, but one such mechanism should be demonstrated).

3. Implementations must send RTCP packets containing an SDES CNAME in every reporting interval.

4. Other SDES items must be sent every third interval, with NAME every 7 of 8 times within that slot and any other SDES items cyclically taking up the 8th slot.

5. Interoperable selection of ‘pass phrase’ for encryption and exchange of media using DES encryption. This must include mapping of the pass phrase to the canonical form.

6. Interoperable transport of RTP via unicast UDP

7. Interoperable transport of RTP via multicast UDP

8. Interoperable transport of RTP via TCP using the encapsulation defined in section 7 of [2].
from payload format to payload type number. Accordingly, the majority of the work needed to demonstrate interoperability consists of testing that media data is exchanged in an interoperable manner using the full range of codecs enumerated in the profile.

The following codecs are assigned static payload types. It should be verified that interoperable implementations exist for each static payload type:

1. The 8kHz PCMU codec (payload type 0)
2. The 8kHz 1016 codec (payload type 1)
3. The 8kHz G726-32 codec (payload type 2)
4. The 8kHz GSM codec (payload type 3)
5. The 8kHz G723 codec (payload type 4)
6. The 8kHz DVI4 codec (payload type 5)
7. The 16kHz DVI4 codec (payload type 6)
8. The 8kHz LPC codec (payload type 7)
9. The 8kHz PCMA codec (payload type 8)
10. The 8kHz G722 codec (payload type 9)
11. The 44.1kHz stereo L16 codec (payload type 10)
12. The 44.1kHz mono L16 codec (payload type 11)
13. The 8kHz QCELP codec (payload type 12)
14. The 8kHz CN codec (payload type 13)
15. The MPA codec (payload type 14)
16. The 8kHz G728 codec (payload type 15)
17. The 11.025kHz DVI4 codec (payload type 16)
18. The 22.050kHz DVI4 codec (payload type 17)
19. The 8kHz G729 codec (payload type 18)

The following codecs use dynamic payload types. It should be verified that some non-RTP means can be used to assign a dynamic payload type to be used by implementations, and that those implementations can then interwork.

1. The GSM-HR codec
2. The GSM-EFR codec
3. The L8 codec
4. The RED codec
5. The VDVI codec

Similarly, interoperable implementation of the following video codecs with static payload type numbers must be demonstrated:

1. The CelB codec (payload type 25)
2. The JPEG codec (payload type 26)
3. The nv codec (payload type 28)
4. The H261 codec (payload type 31)
5. The MPV codec (payload type 32)
6. The MP2T codec (payload type 33)
7. The H263 codec (payload type 34)

The following codecs use dynamic payload types. It should be verified that some non-RTP means can be used to assign a dynamic payload type to be used by implementations, and that those implementations can then interwork.

1. The BT656 codec
2. The H263-1998 codec
3. The MP1S codec
4. The MP2P codec
5. The BMPEG codec

Implementations must also demonstrate interoperable use of the marker bit. That is, the M bit is set on the first packet of each talkspurt for audio and on the last packet of each frame for video.

3 Author’s Address

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4 References

