Application Note

How to stream AAC+ in RTP from Digigram IQOYA to Barix Streaming Client

1 AAC+ in RTP Stream

Barix Streaming Client can decode AAC+ audio from a file (local or remote) as well as within a Shoutcast/Icecast stream.

The firmware, running on AAC+ decoding enabled HW, can also decode AAC+ audio format within a RTP stream under specific configuration for encapsulation, codec parameters and payload type.

Both the AAC (AAC LC) and AAC+ (HE-AAC) are supported. In AAC+ both SBR (v1) and PS (v2) are supported. Nevertheless it has to be noted that depending on the Encoder settings (sampling rate, compression ...) in our device some of the parametric functions of the High Efficiency AAC v2 Profile (SBR, PS) might not be resolved during decoding. More specific: the Parametric Stereo and the Spectral Band Replication are dynamically removed when the DSP is overloaded. As effect in the decoded audio you loose the Stereo effect and the high frequency range.

In order to optimize audio quality for your solution we do recommend to use MONO where possible (e.g. for in-store background music) and lower sampling rate (e.g. 32KHz instead of 48KHz). In this way the stream bit-rate is used at its best giving the highest perceivable audio quality.

Using anything beyond 32khz sampling rate (such as 44.1k or 48k) will just increase the need for bandwidth, or lower the quality when using the same stream bit-rate (e.g. 64kbps). If the application does not requires high audio frequencies (decoded audio frequency is circa half of the sampling rate), as our human ears do not hear much higher than 15kHz, encoding at 32kHz is the highest sampling rate needed to deliver the full-audible spectrum to humans.

Specifically Barix has validate the interoperability of AAC+ in RTP stream with the Digigram IQOYA*Link encoder.
2 Digigrum IQOYA configuration

2.1 System

The IQOYA*Link can be accessed through the WEB https interface (see IQOYA Manual).

Check that your IQOYA is enabled for AAC+

2.2 Audio Settings

Browse “System” – “Audio Setup”

And configure Audio mode = Mono
Sampling Frequency = 32'000 (Hz)

Finally APPLY the changes.
2.3  SEND configuration

Click on the antenna icon to go to the SEND tab
Click the Name field in the SOURCE section and then the CONFIGURATION button.
Now you can edit the DETAILS:
- give a name
- select Type = Analog
- select Mode = Mono

Click the Name field in the DESTINATION section and then the CONFIGURATION button. Now you can edit the DETAILS:
- give a name
- select Type = RTP stream
- give a destination IP address / port
- select “Sound format” = AAC-4_LC+SBR_ADTS 32000Hz Mono 32000b/s
- change Payload to 126

Finally APPLY the changes

NOTE: you cannot save the DESTINATION settings before setting the SOURCE
3 Streaming Client configuration

On the Streaming Client firmware WEB User Interface simply change the receiving RTP port to the one defined as destination on the IQOYA. In this example:

URL1 = rtp://0.0.0.0:9060