Streaming over RTP with Barix products
Barix Application Note

About RTP and lost sound replacement

RTP means Real Time Protocol . The RTP is a stream of UDP frames that apart from audio content contain a small header with time sequence information. This information is used by the receiving party (e.g. Exstreamer) to detect and replace lost frames.

If the Exstreamer detects lost frame(s) then it will replace the missing audio information with repetition(s) of the next incoming frame. This method is used in mobile phones as well.
The sound quality is good because the replacement is usually short and fits well into it's surrounding. Even with massive packetloss, music and speech are still intelligible. The advantage of RTP is that the rhythm of music or speech can be preserved even when packets are lost during transmission.

RTP on the Exstreamer also replaces the old SYNC feature. When the RTP TX port is configured in the setup, the Exstreamer will broadcast the incoming stream with the RTP header in the local network . Synchronization information will be broadcast one port higher in RTCP format.

RTP is an Internet standard, therefore it is interoperable with a wide range of other RTP supporting applications and hardware devices.

Required firmware

The use of RTP with Barix products requires downloading the following firmware from
http://www.barix.com

Instreamer : VB2.02 or higher,
http://www.barix.com/index.php?option=com_docman&task=cat_view&gid=108&itemid=0

Exstreamer: VB8.00 or higher,
http://www.barix.com/index.php?option=com_docman&task=cat_view&gid=102&itemid=0

Barix Streaming Client : VB1.00 or higher (available on request)

Overview of covered scenarios

RTP can be sent by the Instreamer and an incoming stream can be retransmitted as RTP by the Exstreamer. RTP can be received by the Exstreamer or a player program on a PC. Other hardware and software sources and receivers of RTP exist as well.

We will cover the following combinations:

1. Instreamer -> Exstreamer
2. Instreamer -> multiple Exstreamers
3. Multiple Exstreamers playing synchronously
4. Instreamer -> 1 Windows PC (MPEG)
5. Instreamer -> 1 Linux PC (MPEG)
6. Instreamer -> 1 Linux PC (G.711)
7. Instreamer -> Multiple PC's
8. Exstreamer as a small RTP MP3 transmitter

9. Existing RTP stream -> Exstreamer

1. Streaming RTP: Instreamer -> Exstreamer

Instreamer:
AUDIO -> Bit Reservoir Mode : kept empty
STREAMING -> Stream to : RTP, IP of the Exstreamer, 4444

Exstreamer:
STREAMING -> Mode: "4 Streaming Receiver"
STREAMING -> SYNC Port : 4444
STREAMING -> UDP Start Threshold: 3192 (default value)

Note: Try a higher Start Threshold to compensate for jitter in clogged network, but this increases audio delay. Try a lower number if you want low audio delay, but this makes the reception sensitive to network delivery jitter. Min. recommended value is around 2000, max. value is 65535.

Note: If UDP port 4444 clashes with something else, use different number in Instreamer and Exstreamer. If the Exstreamer is behind a NAT ("home/office router"), use NAT's outside IP address in the Instreamer and configure the NAT to forward UDP port 4444 to the Exstreamer.

Tip: You can change AUDIO -> Encoding + Frequency in the Instreamer between MPEG, u-Law 8kHz and A-Law 8kHz. The Exstreamer will switch the format automatically.

2. Streaming RTP: Instreamer -> multiple Exstreamers on a LAN

Proceed as above in "Instreamer -> Exstreamer", but instead of "IP of the Exstreamer", configure 0.0.0.0 into the Instreamer.

Note: the Extstreamers may play out of sync. If you want them to play in sync (e. g. having them in a single room), then see the next chapter.

3. Setting Exstreamers on a LAN to play synchronously

One Exstreamer (let's call it the master) will receive audio from an external source – UDP, TCP, zServer, HTTP server or RTP. The Exstreamer will, regardless of source type, send the received data again to the other exstreamers (slaves) together with synchronization information. Thanks to the sync information they will play acoustically in sync. The traffic between master and slaves is broadcast on UDP port 5555 (RTP - audio) and 5556 (RTCP – sync information). The RTCP transmission is specific to this scenario and is not used in other RTP scenarios.

Master Exstreamer
STREAMING -> RTP TX Port: 5555

Slave Exstreamers:
STREAMING -> Mode : "4 Streaming Receiver"
STREAMING -> SYNC Port: 5555

Note: the 5555 has to be changed if 5555 or 5556 UDP port clash with your existing network traffic.

Note: make sure the audio going into the master Exstreamer is MPEG format. G.711 is not supported in this scenario.

4. Streaming MPEG over RTP: Instreamer -> 1 Windows PC

Note: Some software tools don't support lost sound replacement. If there is a packetloss you will hear it as interrupted sound.

Instreamer:
AUDIO -> Bit Reservoir Mode : kept empty
STREAMING -> Stream to : RTP, IP of the PC, 4444
AUDIO -> Encoding + Frequency: MPEG according to your bitrate preference

Note: If UDP port 4444 clashes is already used by existing traffic, use a different number. If the PC is behind a NAT ("home/office router"), configure NAT's outside IP address into the Instreamer instead and configure the NAT to forward UDP port 4444 to the PC.

VLC

http://www.videolan.org/vlc/
Go into File -> Open Network Stream -> Network and select UDP/RTP, port 4444 and turn IPv6 off. Then press OK.

Winamp version 2 (2.6 or later) or 5. Not Winamp 3.

http://www.winamp.com/
Winamp needs the plugin in_rtp.dll to play RTP. Download it from http://www.live555.com/multikit/winamp-plugin.html into your Winamp "Plugins" folder - usually "C:\Program Files\Winamp\Plugins".
Start Winamp, File -> Play URL and enter `rtp://0.0.0.0:4444`

Zinf (former FreeAmp)

http://www.zinf.org/

FreeMP3 Player

http://www.freewirep2p.com/ (download not available at publication time)
Go into Files and type `rtp://0.0.0.0:4444` into „URL". Then press Open URL.

5. Streaming MPEG over RTP: Instreamer -> 1 Linux PC

Note: If the software refuses to play at all it may be caused by another media player blocking the sound card. Some software tools don't do lost sound replacement. If there is a packetloss you will hear it as interrupted sound.

Instreamer:
AUDIO -> Bit Reservoir Mode : kept empty
STREAMING -> Stream to : RTP, IP of the PC, 4444
AUDIO -> Encoding + Frequency: MPEG according to your bitrate preference

Note: If UDP port 4444 is already used by other traffic, use a different number. If the PC is behind a NAT ("home/office router"), configure NAT's outside IP address into the Instreamer instead and configure the NAT to forward UDP port 4444 to the PC.

Mplayer

http://www.mplayerhq.hu/
and VLC
http://www.videolan.org/vlc/
With a text editor, create a file mplayer.sdp with the following 2 lines:
```
c=IN IP4 0.0.0.0
m=audio 4444 RTP/AVP 14
```
Now run `mplayer sdp://mplayer.sdp` or `vlc sdp://mplayer.sdp`

Zinf (formerly Freeamp)
http://www.zinf.org/
run `zinf rtp://0.0.0.0:4444`

6. Streaming G.711 over RTP: Instreamer -> 1 Linux PC
Note: G.711 sound has telephone quality

Note: If the software refuses to play at all it may be caused by another media player blocking the sound card or by a running sound server. Some software tools don't support lost sound replacement. If there is packet loss you will hear it as interrupted sound.

Instreamer:
STREAMING -> Stream to : RTP, IP of the PC, 4444
AUDIO -> Encoding + Frequency: "uLaw / 8kHz (G.711)" or "aLaw / 8kHz (G.711)"

Note: If UDP port 4444 clashes with something else, replace 4444 in the configuration with a different number which is not odd. If the PC is behind a NAT ("home/office router"), configure NAT's outside IP address into the Instreamer instead and configure the NAT to forward UDP port 4444 to the PC.

Mplayer
http://www.mplayerhq.hu/
and VLC
http://www.videolan.org/vlc/
With a text editor, create a file mplayer.sdp with the following 2 lines:
```
c=IN IP4 0.0.0.0
m=audio 4444 RTP/AVP 8
```
If you have selected u-Law on the Instreamer, replace the number 8 with 0. Now run mplayer:
```
mplayer sdp://mplayer.sdp
```

RTP Tools and ALSA aplay
You need to download and install RTP Tools from http://www.cs.columbia.edu/IRT/software/rtptools/ and have Advanced Linux Sound Architecture installed http://www.alsa-project.org/ which contains the aplay program.

if you selected uLaw:
```
rtpdump -F payload /4444 | aplay -r 8000 -c 1  -f MU_LAW
```
if you selected aLaw:
```
rtpdump -F payload /4444 | aplay -r 8000 -c 1  -f A_LAW
```

Note: if you can't use the aplay program (e. g. you are using OSS instead of ALSA), try to find a different program. Configure it for 8kHz, mono, and u-Law or A-Law, respectively.

7. Streaming RTP: Instreamer -> Multiple PC's on a LAN

Proceed as with single Linux PC, but configure 0.0.0.0 into the Instreamer instead of PC's or NAT's IP address.

8. Turning the Exstreamer into a small RTP MP3 transmitter

First, configure the Exstreamer the way you normally configure it to play your favourite music. You can use zServer or HTTP server with playlists or receive external Internet radio.

In the Exstreamer, set Config -> Settings -> Streaming -> RTP TX port to 4444. Now the content you can hear on the Exstreamer's audio output is broadcast over your LAN on port 4444 using the RTP protocol.

Note: If UDP port 4444 is already used by other traffic, replace 4444 in the configuration with a different number.

Finally set up your PC clients according to “Streaming MPEG over RTP: Instreamer -> 1 Windows PC” or “Streaming MPEG over RTP: Instreamer -> 1 Linux PC” scenarios.

Note: the RTP protocol allows the MPEG frames transmitted to differ in bitrate and other parameters. Some PC media player applications have problems with this. The problem manifests itself as skipping sound, bubbling noises etc. The parameter changes occur during transitions between songs and in VBR encoded mp3 files.

9. Playing existing RTP stream on Exstreamer

You need a RTP stream that is being delivered to the Exstreamer's IP address – either by network broadcast or unicast. You need to know which UDP port number it's taking place on. The stream must be an audio stream in MPEG, u-Law 8kHz or A-Law 8kHz format.

Exstreamer:
STREAMING -> Mode: “4 Streaming Receiver”
STREAMING -> SYNC Port : port number of your stream
STREAMING -> UDP Start Threshold: 3192 (default value)

Note: Try higher Start Threshold to compensate jitter in clogged network, but this increases audio delay. Try lower number if you want low audio delay, but this makes the reception sensitive to network delivery jitter. Min. recommended value is around 2000, max. value is 65535.