

# Internet & Multimedia

## Exercises: using GStreamer

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# GStreamer concepts

## gstreamer tools

gst-inspect  
gst-launch  
gst-editor

media player

VoIP & video conferencing

streaming server

video editor

(...)

## multimedia applications



**protocols**

- file:
- http:
- rtsp:
- ...

**sources**

- alsa
- v4l2
- tcp/udp
- ...

**formats**

- avi
- mp4
- ogg
- ...

**codecs**

- mp3
- mpeg4
- vorbis
- ...

**filters**

- converters
- mixers
- effects
- ...

**sinks**

- alsa
- xvideo
- tcp/udp
- ...

3rd party plugins

## gstreamer plugins

gstreamer includes over 250 plugins

# GStreamer concepts

Gstreamer and input/output: sources and sinks

files – input and output files

input devices, e.g. webcam; output devices, e.g. speaker, screen

network protocols: UDP, TCP, RTP/UDP, RTP/RTCP/UDP, ..



Gstreamer pipeline

# GStreamer : source file & decoding

First example:

```
gst-launch-0.10 filesrc location=hello.ogg !  
oggdemux ! vorbisdec ! alsasink
```



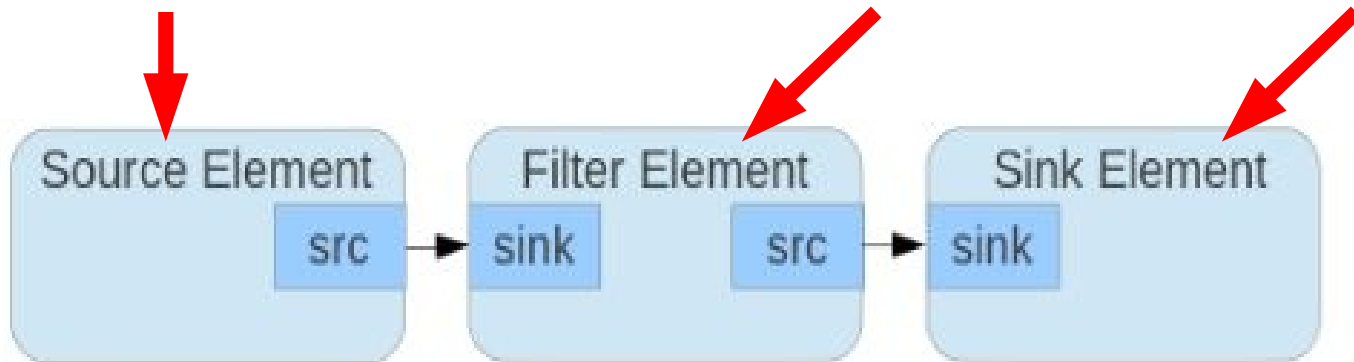
**Exercise:** write the similar pipeline that decodes and plays an mp3 file. The mp3 decoder is called mad. (no **demux** needed), mp3 file has only audio track.

# GStreamer : source file & decoding

Second example:

```
gst-launch-0.10
```

```
filesrc location=hello.ogg ! oggdemux ! vorbisdec !  
audioconvert ! audioresample ! autoaudiosink
```



**Exercise:** write the similar pipeline that decodes and plays an mp3 file with **resample**. The mp3 decoder is called **mad**.



# Decoding & coding

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Third example:

```
gst-launch-0.10 -v filesrc location=Coffee.mp3 !  
mad ! audioconvert ! audioresample !  
vorbisenc ! oggmux ! filesink location=Coffee.ogg
```

**Exercise:** Draw the scheme of the pipeline and write a similar pipeline that decodes an **ogg** file and codes it as **mp3** file. The **mp3** encoder is called **lamemp3enc**.



# Decoding with decodebin

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Fourth example:

```
gst-launch-0.10 filesrc location=hello.ogg !  
decodebin ! autoaudiosink
```

**decodebin** finds automatically the  
necessary elements – **oggdemux** and  
**vorbisdec**



# Decoding and coding

Fifth example – pipeline with 2 threads:

```
gst-launch filesrc location="video.mp4" ! decodebin  
name=d \  
d. ! queue ! audioconvert ! audioresample !  
vorbisenc ! oggmux name=mux \  
d. ! queue ! theoraenc ! mux. \  
mux. ! filesink location="out.ogv"
```

**Exercise:** Draw the scheme of the pipeline and write a similar pipeline that decodes an **ogv** file and codes it as **mp4** file. **Theora** decoder is called **theoradec**, **Vorbis** decoder is called **vorbisdec**. The **mp3** encoder is called **lamemp3enc**. The **h264** encoder is called **x264enc**.





# Using webcam

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Sixth example :

```
gst-launch-0.10 -v v4l2src ! video/x-raw-  
yuv,width=320,height=240 ! ffmpegcolorspace !  
autovideosink
```

**Exercise:** Draw the scheme of the pipeline and write a similar pipeline that creates a smaller (**160x120 pixels**) image.



# Using webcam

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Seventh example :

```
gst-launch-0.10 v4l2src ! 'video/x-raw-yuv,  
width=640,height=480,framerate=30/1' !  
theoraenc bitrate=256 !  
oggmux ! filesink location=webcam.ogg
```

**Exercise:** Draw the scheme of the pipeline and write a similar pipeline that creates an **avi** file with **h264** video encoder. Use **x264enc** encoder and **avimux** container creator. Comment on **bitrate** value !



# Using audio with webcam

Eight example :

```
gst-launch-0.10 -v alsasrc device=hw:1,0 !  
audioconvert ! 'audio/x-raw-  
int,rate=48000,channels=1' ! autoaudiosink
```

**Exercise:** Draw the scheme of the pipeline and write a similar pipeline that creates an audio file (mp3)



# Sending media over UDP protocol

Ninth example :

```
gst-launch-0.10 filesrc location=$1 ! queue !  
udpsink host=localhost port=9999
```



```
gst-launch-0.10 udpsrc port=9999 ! queue !  
decodebin ! audioconvert ! autoaudiosink
```

**Exercise:** Draw the corresponding pipelines and write a similar example that sends the audio to external host (use an IP address). Note the presence of **queue** that operates as buffer.



# Sending media over TCP protocol

Tenth example :

```
#the client of tcpserver - to be launched first !  
echo 'waiting for the connection from server'  
gst-launch-0.10 tcpserversrc host=172.19.64.141  
port=5002 ! decodebin ! autoaudiosink
```

```
gst-launch-0.10 -v alsasrc device=hw:1,0 !  
audioconvert ! 'audio/x-raw-int,  
rate=48000,channels=1' ! speexenc ! queue !  
tcpserversink host=172.19.64.141 port=5002
```

**Exercise:** Explain the differences between UDP and TCP solutions !

# Sending media over RTP/UDP

Eleventh example :

```
gst-launch -v v4l2src ! 'video/x-raw-  
yuv,width=320,height=240' ! jpegenc ! rtpjpegpay ! udpsink  
host=172.19.64.141 port=5011
```

```
gst-launch udpsrc port=5011 \  
caps="application/x-rtp, media=(string)video,  
clock-rate=(int)90000, encoding-name=(string)JPEG,  
encoding-params=(string)1, payload=(int)26" ! \  
gstrtpjitterbuffer ! rtpjpegdepay ! jpegdec ! autovideosink
```

**Exercise:** Explain the usage of **caps** parameters and their relation to RTP protocol



# Sending media over RTP/UDP

Twelve example :

```
gst-launch -v alsasrc device=hw:1,0 ! audioconvert !  
audioresample ! \  
'audio/x-raw-int,rate=16000,width=16,channels=1' ! speexenc !  
rtpspeexpay ! udpsink host=172.19.64.141 port=5001
```

```
gst-launch udpsrc port=5001 caps="application/x-rtp,  
media=(string)audio, clock-rate=(int)16000,  
encoding-name=(string)SPEEX, encoding-params=(string)1,  
payload=(int)110" ! gstrtpjitterbuffer name=jbuf latency=70  
drop-on-latency=True ! rtpspeexdepay ! speexdec !  
audioconvert ! audioresample ! volume volume=10 !  
autoaudiosink
```



# Summary

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