

流媒体高级编程

STREAMING MEDIA DAY05



		09:00 ~ 09:30	作业讲解和回顾
	۲.	09:30 ~ 10:20	
		10:30 ~ 11:20	屏幕流录制
		11:30 ~ 12:20	
		14:00 ~ 14:50	
	下在	15:00 ~ 15:50	混合流录制
	PŦ	16:00 ~ 16:50	
		17:00 ~ 17:30	总结和答疑
+			











- 录制屏幕流
 - 抓取视频显示器上的动态影像保存到本地或推送至远程
 - ✓ 本地录制: ScreenRecorder screen.flv
 - ✓ 推送直播: ScreenRecorder rtmp://192.168.1.166/live/1

















av_gettime



- 获取当前系统时间
 - #include <libavutil/time.h>

int64_t av_gettime (void);

- 返回始自1970年1月1日0点0分0秒,直到函数被调用时的 总微秒数,1微秒=10-6秒,即百万分之一秒



ScreenRecorder

【参见: FFmpeg/Primer/ScreenRecorder】

- 课堂练习
- 录制屏幕流
 - 抓取视频显示器上的动态影像保存到本地或推送至远程
 - ✓ 本地录制: ScreenRecorder screen.flv
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需求分析



- 录制音视频混合流
 - 在Windows上列表显示所有的Video For Windows设备
 和DirectShow设备及其选项
 - 通过用户选择的采集设备, 捕获音视频数据, 保存或推送
 - 分别在独立的线程中,以并发的方式,采集音频和视频流, 避免因捕获过程的相 互等待,丢失帧数据
 - 借助独立的混流线程,
 将捕获到的音视频帧,
 按解码时间戳的升序,
 依次写入目标格式中













知识讲解







av_compare_ts

• 比较时间戳

- #include <libavutil/mathematics.h>

int av_compare_ts (

int64_t ts_a, // 时间戳a
AVRational tb_a, // 时间戳a的单位
int64_t ts_b, // 时间戳b

AVRational tb_b); // 时间戳b的单位

若时间戳a早先于时间戳b,则返回-1;
 若时间戳a迟晚于时间戳b,则返回1;
 若时间戳a同时于时间戳b,则返回0



AVRecorder

【参见:FFmpeg/Primer/AVRecorder】

• 录制音视频混合流

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- 通过用户选择的采集设备, 捕获音视频数据, 保存或推送
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- 借助独立的混流线程,将捕获到的音视频帧,按解码时间 戳的升序,依次写入目标格式中





基于Nginx+RTMP的流媒体系统(下) Tare



- \$ sudo service nginx start
 或
 - \$ sudo service nginx restart
- 14.在服务器上更改防火墙,允许1935/tcp端口
 - \$ sudo ufw allow 1935/tcp
- 15. 在服务器上用netstat检查端口侦听情况
 - \$ netstat –ltn

Proto	Recv-Q	Send-Q	Local	Address	Foreign	Address	State
tcp	0	0	0.0.0	.0:1935	0.0.0:	*	LISTEN
tcp	0	0	0.0.0	.0:8080	0.0.0:	*	LISTEN

基于Nginx+RTMP的流媒体系统(下) Tarena

16.在服务器上通过浏览器检查nginx服务状态

- http://localhost:8080
- http://localhost:8080/stat

🛞 🖨 🗊 Welcome to nginx! - Mozil	la Firefox			
Welcome to nginx! × +				
(ilocalhost:8080	C Q	. 搜索	»	=

Welcome to nginx!

If you see this page, the ngi Further configuration is requ	RTMP statist	P statistics lics lost:8080/s ¹	x - Mozill	la Firefo 	x					C	Q .搜索		\$	ê.	↓ ^		=
For online documentation a Commercial support is avai	RTMP	#clients		Video	ť.	0	Aud	io		In bytes	Out bytes	In bits/s	Out bits/	s State	e Time		
Commercial cappercie ara	Accepted: 0		codec	bits/s	size 1	fps code	bits/s	freq	chan	0 KB	0 KB	0 Kb/s	0 Kb/s		7m 51	s	
Thank you for using nginx.	vod																
	vod streams	0															
	live																
	live streams	0															
	hls																
	live streams	0															
	Generated by	nginx-rtm	np-mod	ule 1.1.4	1, ngin:	<u>x</u> 1.7.5, pic	1077, b	uilt Ma	r 17 20	017 20:17	:33 gcc 5.4.0	20160609) (Ubuntu !	5.4.0-6u	buntu1^	16.04	.4)

基于Nginx+RTMP的流媒体系统(下) Tarena

17.在服务器上将视频点播文件(1.mp4 ...)拷贝到~/Videos 目录下,保证任何用户对其可读

🛞 🗇 🗊 minwei@ubuntu: ~/Videos				
<pre>minwei@ubuntu:~/Videos\$ pwd</pre>				
/home/minwei/Videos				
<pre>minwei@ubuntu:~/Videos\$ ls -l</pre>				
总用量 3135604				
-rw-rr 1 minwei minwei 417170370	8月	12	2014	1.mp4
-rw-rr 1 minwei minwei 297089682	8月	12	2014	2.mp4
-rw-rr 1 minwei minwei 373161819	8月	12	2014	3.mp4
-rw-rr 1 minwei minwei 361059118	8月	12	2014	4.mp4
-rw-rr 1 minwei minwei 423277865	8月	12	2014	5.mp4
-rw-rr 1 minwei minwei 960192458	8月	15	2014	6.mp4
-rw-rr 1 minwei minwei 169194964	1月	24	21:20	7.mp4
-rw-rr 1 minwei minwei 109132009	11月	7	2015	8.mp4
-rw-rr 1 minwei minwei 100544991	1月	22	19:05	9.mp4
minwei@ubuntu:~/Videos\$				

基于Nginx+RTMP的流媒体系统(下) Tarena

18.测试点播媒体文件

- 播放端需要先从http://www.videolan.org下载并安装
 VLC media player
- 在播放端用VLC media playe打开网络串流: rtmp://192.168.232.130/vod/1.mp4



基于Nginx+RTMP的流媒体系统(下) Tare

19.测试RTMP直播媒体文件

- 推送端需要先从http://ffmpeg.org下载并安装FFmpeg
- — 在推送端用FFmpeg推送媒体流:

 ffmpeg -re -i 1.mp4 -vcodec libx264 -vprofile
 baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv s 720x480 -q 10 rtmp://192.168.232.130/live/1
- 在播放端用VLC media player打开网络串流: rtmp://192.168.232.130/live/1



基于Nginx+RTMP的流媒体系统(下) Tare



- 在推送端用FFmpeg推送媒体流:
 - ffmpeg -re -i 1.mp4 -vcodec libx264 -vprofile baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv
 - s 720x480 -q 10 rtmp://192.168.232.130/hls/1
- 在播放端用VLC media player打开网络串流:
 rtmp://192.168.232.130/hls/1



基于Nginx+RTMP的流媒体系统(下) Tarer

- 21.测试RTMP直播摄像头和麦克风
 - 在推送端用FFmpeg推送媒体流:
 - ffmpeg -f dshow -i video="BisonCam, NB Pro" -f dshow -i audio="麦克风 (Realtek High Definition Audio)" -pix_fmt yuv420p -vcodec libx264 -vprofile baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv s 640x480 -q 10 rtmp://192.168.232.130/live/1
 - 在播放端用VLC media player打开网络串流:
 rtmp://192.168.232.130/live/1



基于Nginx+RTMP的流媒体系统(下) Tar

22.测试HLS直播摄像头和麦克风

- 在推送端用FFmpeg推送媒体流:
 - ffmpeg -f dshow -i video="BisonCam, NB Pro" -f dshow -i audio="麦克风 (Realtek High Definition Audio)" -pix_fmt yuv420p -vcodec libx264 -vprofile baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv s 640x480 -q 10 rtmp://192.168.232.130/hls/1
- 在播放端用VLC media player打开网络串流:
 rtmp://192.168.232.130/hls/1

🖬 命令提示符 - ffmpeg -f dshow -i video="BisonCam, NB Pro" -f dshow -i audio="麦克风 (Realtek High Definition Audi 🗕 🗆 🗙	
[dshow @ 000000000055ce00] real-time buffer [BisonCan, NB Pro] [video input] too full or near too full (101% of size: 30	^
(LZD) [rtoursize parameter]) [trame aropped]	
Last Message repeated (times	
Tibezov s coccocceccon so) -gecale is ignored, -cri is recomended.	
interact a conconception with the philips of the same the same the same the same the same the same and the same	
1990 Fithefaire avenue will five descend	
Liby204 8 00000000800dfa01 profile Constrained Baseline level 3 0	
libx264 0 0000000008064501 264 - core 148 r2714 b97a00 - H.264/WFEG-4 AVC codec - Corvleft 2003-2016 - http://www.vid	
and an org/x204, html - ontions: cabac=0 ref=3 deblock=1:0:0 analyse=0x1:0x111 me=hex subme=7 nsy=1 nsy rd=1.00:0.00 nixed	
ref=1 me range=16 chrona me=1 trellig=1 8x8dct=0 con=0 deadzone=21.11 fast pskip=1 chrona op offset=-2 threads=12 looka	
head threads=2 sliced threads=0 nr=0 decimate=1 interlaced=0 bluray compat=0 constrained intra=0 bframes=0 weightp=0 key	· •
int=250 kevint min=25 scenecut=40 intra refresh=0 rc lookahead=40 rc=crf mbtree=1 crf=23.0 gcomp=0.60 gpmin=0 gpmax=69 g	· · · · · · · · · · · · · · · · · · ·
pstep=4 ip ratio=1.40 ag=1:1.00	
Output #0, flv, to 'rtmp://192.168.232.130/hls/1':	
Metadata:	. 5
encoder : Lawf57.50.100	1
Stream #0:0: Video: h264 (libx264) ([7][0][0][0] / 0x0007), yuv420p, 640x480, q=-11, 30 fps, 1k tbm, 30 tbc	1
Metadata:	1
encoder : Lavc57.64.101 libz264	
Sade data:	
cpb: bitrate max/min/avg: 0/0/ butter size: 0 vov delay: -1	
Stream Hori: Audio: and (LC) ([10][0][0] / 0x000k), 40000 Hz, stereo, titp, 120 kb/s	
setsors:	
encouer : Laveor. 04.101 aac	
Stran $B(0) = 2000$ (rampidae (nativa) $\rightarrow 2024$ (libv264))	
Stram #1:0 -> #0:1 (new slobe (netro) -> asc (netro))	
Press [a] to ston [2] for baln	
frame= 112 fps= 32 g=29.0 size= 368kB time=00:00:02.95 hitrate=1019.7kbits/s smeed=0.838x	~



基于Nginx+RTMP的流媒体系统(下) Tarer

- 23.测试RTMP直播屏幕和麦克风
 - 在推送端用FFmpeg推送媒体流:
 - ffmpeg -f gdigrab -i desktop -f dshow -i audio="麦 克风 (Realtek High Definition Audio)" -pix_fmt yuv420p -vcodec libx264 -vprofile baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv -s 640x480 -q 10 rtmp://192.168.232.130/live/1
 - 在播放端用VLC media player打开网络串流: rtmp://192.168.232.130/live/1

🖬 命令揭示符 - ffmpeg -f gdigrab -i desktop -f dshow -i audio="麦克风 (Realtek High Definition Audio)" -pix_fmt yuv42 📃 🗌	×
Stream #0:0: Video: bmp, bgra, 1920x1080, 1988680 kb/s, 29.97 fps, 1000k tbr, 1000k tbn, 1000k tbc	^
Guessed Channel Layout for Input Stream #1.0 : stereo	
input #1, dshow, trom audio=42 mmtr/(Kealtek High Definition Audio) :	
Duration: M/A, start: 391014.129000, Ditrate: 1411 RD/S	
STream HIO: Audio: pcm_siole, 44100 Hz, steree, sio, 1411 RD/s	
[15:204 0 000000000ccesa0] -gscale is ignored, -crf is recommended.	
libx204 8 000000000eceaa01 using cpu capabilities: MMA2 5582Fast 55585 5584.2 AVA FMA5 AVA2 L2CM1 BM12	
librazof @ 0000000000ceceas0] protile Constrained Baseline, level 5.0	
[1182204 @ 000000000000000000000000000000000	
eoran, org/x204, html - oprions; cabac-o ret-3 decidex-110:0 analyse-ori:0xill me-nex subme-1 psy-1 psy-1 decide on in	ard .
ret-1 me_range-10 chroma_me-1 trellis-1 oxodct-0 cqm-0 deadzone-21,11 tast_psg1p-1 chroma_qp_ottset2 threads-12 10	220
nest inreads-2 sliced inreads-0 nr-0 decimate-1 interlaced-0 bluray compat-0 constrained intra-0 blrames-0 weightp-0 i	
int-250 keyint_min-25 scenedut-40 intra_reiresh-0 re_inoganean-40 re-cri moiree-1 cri-25.0 dcomp-0.00 dpmin-0 dpmax-0	
Determined and a second s	
Watadata	
anodar : Lart57.56 100	
Straam #0:0: Video: 5054 (151x054) ([7][0][0][0][0] / 0x0007) xew420e 640x480 c=-1-1 29 97 for 1k the 29 97 fl	
Watadata	
encoder : Law57 64 101 libr284	
Side data:	
cph: hitrate max/min/avg: 0/0/0 huffer size: 0 yby delay: -1	
Stream #0:1: Audio: and (LC) ([10][0][0][0] / 0x000Å), 48000 Hz, stereo, fltp, 128 kb/s	
Matadata;	
encoder : Lavc57.64.101 aac	
Stream mapping:	
Stream #0:0 -> #0:0 (bmp (native) -> h284 (libx264))	
Stream #1:0 -> #0:1 (pcm_s16le (native) -> aac (native))	
Press [g] to stop, [?] for help	
frame= 174 fps= 20 g=29.0 size= 495kB time=00:00:07.00 bitrate= 578.4kbits/s speed=0.793x	~



基于Nginx+RTMP的流媒体系统(下) Taren

24.测试HLS直播屏幕和麦克风

– 在推送端用FFmpeg推送媒体流:

ffmpeg -f gdigrab -i desktop -f dshow -i audio="麦 克风 (Realtek High Definition Audio)" -pix_fmt yuv420p -vcodec libx264 -vprofile baseline -acodec aac -ar 48000 -strict -2 -ac 2 -f flv -s 640x480 -q 10 rtmp://192.168.232.130/hls/1

- 在播放端用VLC media player打开网络串流: rtmp://192.168.232.130/hls/1

🖬 命令提示符 - ffmpeg -f gdigrab -i desktop -f dshow -i audio="麦克风 (Realtek High Definition Audio)" -pix_fmt yuv42 🗕 🗆 🗙	
Stream #0:0: Yideo: bmp, bgra, 1920x1080, 1988680 kb/s, 29.97 fps, 1000k tbr, 1000k tbn, 1000k tbc	C.
Trout #1 drhow from 'audio=30 mmac2/Realtak High Definition Audio)':	
Duration: W/A start: 391674 (19900) bitvata: 1411 bb/s	
Stream #1:0: Audio: nom sifele 44100 Hr. stereo sife 1411 kb/s	
[]ibx264 0 000000000eceas0] _oscale is ignored _outf is recommended.	
libx264 @ 000000000eceaa0] using cru canabilities: WWW2 SSE2East SSSE3 SSE4.2 AVX FMA3 AVX2 L2CWT BWT2	
[libx264 @ 000000000eccaa0] profile Constrained Baseline, level 3.0	
libx264 0 000000000eceaa0 264 - core 148 r2744 b97ae06 - H. 264/WPEC-4 AVC codec - Corvleft 2003-2016 - http://www.vid	G
eolan.org/x204.html = options; cabac=0 ref=3 deblock=1:0:0 analyse=0x1:0x111 me=bex subme=7 psy=1 psy rd=1.00:0.00 mixed	0
ref=1 me range=10 chrona me=1 trellis=1 8x8dct=0 cgn=0 deadzone=21.11 fast pskip=1 chrona gp offset=-2 threads=12 looka	
head threads=2 sliced threads=0 nr=0 decinate=1 interlaced=0 bluray compat=0 constrained intra=0 bframes=0 weightp=0 key	6
int=250 kevint min=25 scenecut=40 intra refresh=0 rc lookahead=40 rc=crf mbtree=1 crf=23.0 gcomp=0.60 gpmin=0 gpmar=69 g	
pstep=4 ip ratio=1.40 ao=1:1.00	
Dutput #0, flv, to 'rtmp://192.108.232.130/live/1':	M
Metadata;	
encoder : Lavf57.56.100	
Stream #0:0: Video: h264 (libx264) ([7][0][0][0] / 0x0007), yuv420p, 640x480, g=-11, 29.97 fps, 1k thn, 29.97 tbc	
Metadata:	
encoder : Lavc57.64.101 libz264	
Side data:	1
cpb: bitrate nax/nin/avg: 0/0/0 buffer size: 0 vbv_delay: -1	
Stream #0:1: Audio: aac (LC) ([10][0][0][0] / 0x000A), 48000 Hz, stereo, fltp, 128 kb/s	
Metadata:	€
encoder : Lavc57.64.101 aac	1000
Stream mapping:	
Stream #0:0 -> #0:0 (bmp (native) -> h264 (libx264))	00:
Stream, #1:0 -> #0:1 (pcm_sible (native) -> aac (native))	
Press [g] to stop, [?] for help	
trane= 174 tps= 20 q=29.0 size= 495kB tine=00:00:07.00 bitrate= 578.4kbits/s speed=0.793x	





总结和答疑