

BUUCTF voip

原创

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33 篇文章 1 订阅

订阅专栏

VoIP——基于IP的语音传输（英语：Voice over Internet Protocol，缩写为VoIP）是一种语音通话技术，经由网际协议（IP）来达成语音通话与多媒体会议，也就是经由互联网来进行通信。其他非正式的名称有IP电话（IP telephony）、互联网电话（Internet telephony）、宽带电话（broadband telephony）以及宽带电话服务（broadband phone service）。

[Wireshark抓取RTP包，还原语音](#)

VoIP使用RTP协议对语音数据进行传输，语音载荷都封装在RTP包里面。

法1:

1. RTP——>RTP流

voip.pcap

文件(E) 编辑(E) 视图(V) 跳转(G) 捕获(C) 分析(A) 统计(S) 电话(Y) 无线(W) 工具(I) 帮助(H)

应用显示过滤器 ... <Ctrl-/>

No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	192.168.56.102	192.168.56.101	RTP	1073	Request: 1
2	0.000595	192.168.56.101	192.168.56.102	RTP	632	Status: 40
3	0.011066	192.168.56.102	192.168.56.101	RTP	447	Request: A
4	0.012304	192.168.56.102	192.168.56.101	RTP	1295	Request: 1
5	0.013392	192.168.56.101	192.168.56.102	RTP	431	Status: 10
6	0.102599	192.168.56.101	192.168.56.102	RTP	602	Status: 20
7	0.104286	192.168.56.102	192.168.56.101	RTP	475	Request: A
8	0.382781	192.168.56.101	192.168.56.102	RTP	86	Receiver F
9	0.382813	192.168.56.102	192.168.56.101	RTP	114	Destinatic
10	0.608371	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
11	0.608591	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
12	0.608660	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
13	0.608724	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
14	0.608787	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
15	0.608847	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
16	0.608909	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (
17	0.617431	192.168.56.102	192.168.56.101	RTP	214	PT=ITU-T (

> Frame 1: 1073 bytes on wire (8584 bits), 1073 bytes captured (8584 bits)
> Ethernet II, Src: PcsCompu_55:7f:85 (08:00:27:55:7f:85), Dst: PcsCompu_43:4a:e9 (08:00:27:55:4a:e9)
> Internet Protocol Version 4, Src: 192.168.56.102, Dst: 192.168.56.101
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060

> Session Initiation Protocol (INVITE)

<

0000 08 00 27 43 4a e9 08 00 27 55 7f 85 08 00 45 00 ..'CJ... 'U...E.

2. 点击 分析——>播放流 就可以听到声音

The image shows the Wireshark interface for analyzing an RTP stream. The main window displays a list of packets with columns for No., Time, Source, and Destination. The selected packet (No. 27) is highlighted in red. A detailed view of the selected packet is shown below, including fields like SSRC and payload (g711U, telephone-event). The bottom of the window features playback controls, including a '播放流' (Play Stream) button, which is highlighted with a red arrow. The status bar at the bottom indicates '分组: 2302' and '找到 1 个流'.

No.	Time	Source	Destination
1	0.000000	192.168.56.102	192.168.56.101
2	0.000595	192.168.56.101	192.168.56.102
3	0.011066	192.168.56.102	192.168.56.101

源地址	源端口	目的地址	目的端口	SSRC	载荷	分组	丢弃
192.168.56.101	14728	192.168.56.102	5066	0x599ce685	g711U	1059	1 (0.1%)
192.168.56.102	5066	192.168.56.101	14728	0xe5f34496	g711U, telephone-event	1208	0 (0.0%)

正向	反向	图形				
分组	序列	Delta (ms)	抖动 (ms)	扭曲	带宽	标
19	2847	0.00	0.00	0.00	1.60	
20	2848	19.98	0.00	0.02	3.20	
27	2850	40.01	0.00	0.00	4.80	
28	2851	19.99	0.00	0.01	6.40	
30	2852	19.99	0.00	0.02	8.00	
33	2853	19.99	0.00	0.03	9.60	
34	2854	20.01	0.00	0.02	11.20	
37	2855	20.23	0.02	-0.21	12.80	
39	2856	19.77	0.03	0.02	14.40	
40	2857	20.04	0.03	-0.01	16.00	
43	2858	19.97	0.03	0.01	17.60	
44	2859	19.99	0.03	0.02	19.20	
46	2860	20.03	0.03	-0.00	20.80	
49	2861	19.99	0.03	0.01	22.40	
50	2862	20.01	0.03	0.00	24.00	
52	2863	19.99	0.03	0.01	25.60	
55	2864	19.99	0.02	0.02	27.20	
56	2865	20.03	0.03	-0.01	28.80	
58	2866	19.98	0.03	0.02	30.40	
61	2867	20.07	0.03	-0.06	32.00	
62	2868	19.93	0.03	0.02	33.60	
64	2869	20.02	0.03	-0.01	35.20	
67	2870	19.96	0.03	0.04	36.80	
68	2871	20.03	0.03	0.01	38.40	
71	2872	20.01	0.03	-0.00	40.00	
73	2873	19.97	0.03	0.03	41.60	

或者

法2

直接主菜单点击 电话——>VoIP电话 就可以听到声音