

Expose VoIP Problems Using Wireshark



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Sniff out VoIP problems with Wireshark.

Vendors would have you believe you need to spend thousands of dollars on protocol analyzer software to troubleshoot Voice over IP (VoIP) networks. After all, voice quality suffers if packets are late, missing or out of order. Fortunately, Wireshark is an open-source package that can do everything these tools can and more.

Wireshark, formerly known as Ethereal, is a network protocol analyzer. Its job is to listen to network traffic, display it in a format that makes sense and then help you find problems. VoIP involves a complex set of protocols that Wireshark can decode and relate to each other. For example, the procedure to set up a call involves a different protocol than the voice traffic itself. Wireshark uses the information from the call setup to better understand the voice flow. With this data in your hands, you can isolate the cause of VoIP problems.

To find the cause of VoIP problems, you must be able to follow the flow of calls from start to finish and ensure that the correct events are happening. This requires an understanding of both the underlying protocols and some telephony concepts. Fortunately, Wireshark provides some excellent tools to help interpret the data. The rest of this article focuses on using Wireshark's tools to solve three common VoIP problems.

The first example is of a phone that's not working—every time a number is dialed, the phone idles, and no ringing is heard in the earpiece. Wireshark is used here to look at the traffic between the phone and the PBX.

Launch Wireshark as root, and select Capture→Options to bring up the Capture Options dialog, as shown in Figure 1. Ensure that the correct interface is being used; otherwise, you won't see the traffic. In this example, I am capturing traffic on the PBX's only interface, eth0.

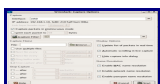


Figure 1. Wireshark Capture Options Dialog Configured for VoIP Analysis

Before you capture, you have the option of specifying a capture filter to limit the number of packets you have to deal with. Because both the signaling traffic (SIP) and voice traffic (RTP) are UDP-based, I specify `udp` as a capture filter. As a convenience, I also check the Update list of packets in real time option so I can verify I'm getting the proper traffic. Click the Start button to start capturing, and soon the upper pane will show several packets (Figure 2).

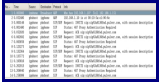


Figure 2. Wireshark Capture of an IP Phone's Traffic

Figure 2 shows the packet list. The first four columns show the packet number, time and the addresses of the hosts involved. Because Wireshark understands SIP, it is able to identify the packets as SIP in the Protocol column and give detailed information in the Info column.

A cursory examination of the information in Figure 2 shows that the same three packets repeat themselves four times. First, the ipphone host sends a SIP INVITE message to pbxhost, which is used to open a conversation. Next, the PBX responds with "407 Proxy Authentication Required" to which the phone responds with an ACK, and then tries again in packet 6. Because this process repeats itself continuously, it's a good guess that the phone is trying to authenticate with the wrong user name and password.

Monitoring: It's All About Location

If some of the packets you are looking for don't reach the network interface of the machine running Wireshark, you've got a problem. Switched networks, by design, try to limit the traffic only to the ports involved with the communication in order to improve performance. Spanning trees and IP routing only complicate the matter of trying to find out where to put your probe.

Your best bet is to capture traffic directly on one of the machines involved. If you can't get Wireshark on that machine (it runs on Windows too), you always can capture traffic locally with `tcpdump` and ship the trace to another machine for analysis. If that isn't possible, or if the traffic spans multiple machines, it's time to get help from your network hardware.

Most managed switches offer a feature that will mirror all the frames from one interface to another. Thus, you could plug your Wireshark station in to a switch port with your Internet connection, mirror all the traffic from the Internet router to your workstation, and you'd be able to analyze all Internet traffic. Cisco calls this Switchport Analyzer or SPAN, HP calls it a Monitor Port, and 3Com calls it a Roving Analysis Port.

The choice of which port to mirror depends on your network, but generally, ports that connect to other switches or routers are a good bet. This is also where capture filters help reduce the background noise you have to deal with.

The first example made use of Wireshark's high-level interpretation of individual packets to help highlight a simple problem. The next example requires a more holistic view of a VoIP call. Here, Wireshark is used to view the SIP call setup and then identify the critical packets. In this example, a phone is experiencing one-way audio. Outbound phone calls ring and are

picked up, but only one party hears audio. The conversation was captured in the same manner as in the previous example, resulting in 140 frames. This is too many to look at without some help, so some of Wireshark's advanced analysis tools are used.

With the capture loaded, select Statistics→VoIP calls. Wireshark analyzes the capture for any VoIP-related packets and provides a summary on the screen. Click on the call (if there are multiple calls shown, hold down the Ctrl key to select additional calls), and click on the Graph button. You will see a summary of the SIP call, as shown in Figure 3.



Figure 3. Wireshark's Graphical Interpretation of a SIP Conversation

The Graph Analysis dialog shows the SIP messages sent by the various parties. The first message in Figure 3 is an INVITE message, which is the first step in setting up a call. In this case, the IP phone is asking the PBX to place a call to a SIP address of sip:613@fwd.pulver.com, which is an echo service provided by Free World Dialup for testing SIP calls. The response to this is a request for authentication, which is acknowledged. The phone tries again and is given a status of "Trying" by the PBX. The PBX then proceeds to INVITE the remote endpoint by contacting fwd.pulver.com. Several more messages are exchanged before the call is set up properly.

The problem at hand is one-way audio. Indeed, the Graph Analysis window shows that the IP phone sent Real-Time Protocol (RTP) voice data to a SIP endpoint on the Internet, but it does not show a stream in the reverse direction. To determine why no audio was sent back, it is necessary to read deeper into the SIP packets.

One of SIP's jobs is to set up the RTP stream between two endpoints. It does this through the Session Description Protocol (SDP), which carries the information about codecs, IP addresses and port numbers that is necessary for VoIP to work. Note that the endpoints speaking SDP need not be the ones talking to each other! In Figure 3, pbxhost and fwd.pulver.com are negotiating with each other over the Internet, but they each specify a different endpoint to terminate the voice call. The RTP streams are unidirectional, so a full duplex conversation requires a separate RTP stream to be set up in each direction, using two separate SDP messages.

With that in mind, it seems prudent to find out the SDP information sent from pbxhost to fwd.pulver.com. This SDP message will contain the IP address, UDP port and codec that the remote end is to use to talk to the local IP phone. SDP messages are tagged with SDP in the Graph Analysis window to help you spot them. This packet is at offset 0.490 in Figure 3. When you click on the message in the Graph Analysis window, the corresponding packet is highlighted in the packet list pane of the main Wireshark window.

Wireshark also decodes the protocols contained within the currently selected packet in the packet detail pane, in addition to providing a summary of the packets in the packet list pane. Decodes are separated for each layer, such as Ethernet, IP, UDP and SIP. You can dig into the fields by clicking the arrows on the left side of the window. Figure 4 shows the SDP message from above with some of the relevant fields expanded.

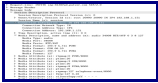


Figure 4. Wireshark's Display of a SIP Packet Containing an SDP Message

Each line in an SDP message describes a particular attribute of the session to be created and follows a simple attribute=value format, where the attribute is a single letter and the value is a text string. Wireshark uses its understanding of the protocol to add some extra text, such as descriptions of the attributes. For VoIP sessions, the important attributes are:

- a—attributes, such as codecs and silence suppression.
- c—connection information, including the IP address expecting the RTP stream.
- m—media description, including the port number on which the RTP endpoint will be listening.

From the Connection Information (c) line, you can determine that the host being sent is 192.168.1.151. The m attribute specifies that the RTP stream is expected to be on port 34008. Looking through the attributes, the PCM u-law and a-law codecs are offered, along with the compressed GSM format and touch tones via the telephone-event media format. The problem with this offering is the IP address 192.168.1.151 is a private address, unreachable from the Internet. When the remote host tries to send packets to 192.168.1.151, the packets are lost because no such route exists on the Internet.

For the sake of completeness, the SDP information for the other side of the conversation is at time 7.554. After the offer is acknowledged, the two phones begin sending voice packets to the address and port specified in the earlier SDP messages. Because pbxhost offered an invalid address to fwd.pulver.com, the other half of the voice call is never seen.

This illustrates a common problem when using VoIP through a NAT gateway. The problem can be solved several ways depending on one's needs, but that's another article!

So far, the troubleshooting has focused on the call signaling aspect of VoIP. Once the endpoints begin sending RTP streams, the effects of network quality can be heard in the audio.

Network conditions that affect VoIP are latency, jitter and packet loss. Latency is the time it takes for a packet to travel from point A to point B. Jitter is the variation in latency over a series of packets. Loss is the number of packets sent from point A that never make it to point B. Because VoIP audio is sent over real-time UDP, a packet that arrives out of order has to be discarded if the packet ahead of it already has been played.

For the final example, I made a call and captured the conversation including the call setup. Wireshark uses the SIP information to get more details about the RTP packet stream, which enables the RTP analysis tools to be used.

Select Statistics→RTP→Show All Streams. Wireshark uses the decoded packets to provide a list of all the audio conversations and some basic statistics, as shown in Figure 5.

Stream	Direction	Protocol	Port	Max Delta	Max Jitter	Packet Loss
1	→	RTP	10000	84ms	11ms	X
2	←	RTP	10000	84ms	11ms	X
3	→	RTP	10000	84ms	11ms	X
4	←	RTP	10000	84ms	11ms	X

Figure 5. Wireshark's List of Found RTP Streams

Figure 5 shows four streams, because each audio stream comprises two streams in opposite directions, and the PBX was bridging the connection between the two phones. The first indication of a problem is the final field, which shows an X when there are VoIP-related problems. The Max Delta (latency) was 84ms for that stream, which is good. The Max Jitter was also good at 11ms (150ms of one-way latency and 20ms of jitter are the limits of what's considered acceptable). However, nearly 20% packet loss was encountered, which is extraordinary!

Select the problematic stream for further analysis by clicking on it. After that, click on the Find Reverse button to select the other half of the conversation. Finally, click on Analyze to provide a packet-by-packet look at the stream. Lost packets will show up as having the wrong sequence number. This screen, shown in Figure 6, also displays helpful statistics, such as the current bandwidth, latency and jitter.



Figure 6. RTP Stream Analysis of a Stream with Excessive Packet Drops

Figure 6 clearly shows some network problems, because the packet loss is not normal. (The steady bandwidth and regularly spaced packet drops are strong indicators of priority queue overruns or policing in network equipment.) This loss will not be seen in the reverse direction, because the capture was taken close to the source of the conversation. You can save the contents of the window with the Save CSV button.

Network problems are more difficult to solve, because they require interaction with network equipment and potentially with other parties. If your VoIP environment doesn't use the Internet, you can capture packets at various points on the network to find where the poor conditions are introduced. If the call flows over the Internet, you must investigate your connection to the Internet and possibly work with your carrier.

The RTP analysis can be used with proprietary systems with some extra configuration. If Wireshark doesn't understand the signaling, it won't be able to decode all of the RTP information. Select Edit→Preferences→Protocols→RTP, and check the Try to decode RTP outside of conversations box. Analysis will be a bit slower, but Wireshark will try to determine whether every UDP packet belongs to an RTP stream, allowing you to use the RTP tools to look at the call.

The final feature we examine here is the ability to listen to the contents of the voice call. With the Stream Analysis window still open, select the Save Payload button, select the .au file format, and provide a filename. After pressing the OK button, the voice call itself is saved to your hard drive. The resulting file can be played in XMMS, soxplay or some other audio program.

Troubleshooting VoIP is unlike most other network troubleshooting, because of the separate signaling and voice protocols, and the real-time nature of voice traffic. These three examples have shown the important features of Wireshark that deal specifically with analyzing the SIP call setup and RTP voice streams.

These features rival those of commercial products and enable to you find the root of VoIP problems.