AudioCodes Quick Reference Guide

Voice Quality Troubleshooting

Background:

This guide will provide a method for examining data obtained when troubleshooting Voice Quality issues.

Tools Required:

The following tools are available for download on the portal.
INI Viewer
Syslog Viewer
Wireshark and Audiocodes Plugins
Note: If you require assistance with installing the plugins, please open a Service Request.
The plugins are built for respective version of Wireshark and should not be used on a different version other that the version specified.
https://www.wireshark.org/download/win32/all-versions/
https://www.wireshark.org/download/win64/all-versions/

Audible and Visual indicators of Voice Quality degradation:

Audible indications:
No audible speech observed. One way audio, Broken, choppy, or garbled audio, etc…

Syslog indicators:
PL:x RTP Packet loss for x number of packets - These packets were not received by the device.
RO:x RTP Reorder - This indicates that he device received packets out of sequence.
RS:x RTP Source is incorrect - The RTP packets are arriving from a different source, then negotiated by SDP.
RP RTP Payload is incorrect - The RTP packets are of a different coder, then negotiated by SDP.
ETC… Reference Syslog Error Name Descriptions in the User's Manual for abbreviations.

Example:
The Syslog message events that the device sends are denoted by unique abbreviations. The following example shows an abbreviated event in a Syslog message indicating packet loss (PL):

Apr 4 12:00:12 172.30.1.14 PL:5 [Code:3a002] [CID:3294] [Time: 20:17:00]

DTMF or RFC2833 indications:
DTMF not being observed or acted upon when digits are entered.
DTMF duplicate digits received.
Differing RFC2833 payloads for DTMF

Payloads:

Reserved/Static payloads are 0-95
Dynamic payloads are 96-127

Payloads continued:

Reserved payloads from RFC 3551(incomplete list) follows:

PT	encoding name	media type	clock rate (Hz)	channels
0	PCMU	A	8,000	1
1	reserved	A		
2	reserved	A		
3	GSM	A	8,000	1
4	G723	A	8,000	1
5	DVI4	A	8,000	1
6	DVI4	A	16,000	1
7	LPC	A	8,000	1
8	PCMA	A	8,000	1
9	G722	A	8,000	1
10	L16	A	44,100	2
11	L16	A	44,100	1
12	QCELP	A	8,000	1
13	CN	A	8,000	1
14	MPA	A	90,000	(see text)
15	G728	A	8,000	1
16	DVI4	A	11,025	1
17	DVI4	A	22,050	1
18	G729	A	8,000	1

RFC2833 payloads use the dynamic payloads:
There is no standard. Most vendors use 101 or 100. You may also see 96, Any dynamic value can be used. The SDP exchange will determine the value used.

Session Description Protocol(SDP):

The SDP exchange is an offer/answer model. The endpoint that originates the call is normally the sends the SDP offer. The terminating side sends its SDP selecting from the offers SDP.
Although there are many instances where the offer may arrive from the terminating side.

When SDP is received in the initial Invite Early Media can be used, but it is not mandatory.
When there is no SDP in the initial Invite, this is called Delayed offer. This guides does not cover these topics.

Session Description Protocol cont.

The C(Connection) line sent in the Offer SDP, indicates the IP addr in which to send RTP.
The M(Media) line indicates: What type of Media (Audio) and port.
What type of Audio RTP/AVP(RTP) or RTP/SAVP(SRTP)
The payloads 0 and 101(g.711 ulaw and RFC2833)
The a(attributes) define the payload:
rtpmap:0 g.711 ulaw
rtpmap:101 1st attribute for RFC2833
rtpmap:101 2nd attribute for RFC2833
ptime:20 (20 msec. if there is no attribute 20 msec is the default)
maxptime:20 (no greater than 20 msec ptime. Some entities send 30 msec. This is not mandatory.

You can have multiple coders and many more parameters. This is a basic example.
Send your RTP 10.10.10.10 port 9386, using g.711 ulaw, with RFC2833 of 101, etc…

Offer

```
v=0
o=- 1568130704 1568130705 IN IP4 10.10.10.10
s=IGW
c=IN IP4 10.10.10.10
t=0 0
m=audio 9386 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=maxptime:20
```

Send your RTP 192.168/1/2 port 7050, using g.711 ulaw, with RFC2833 of 101
In addition provide comfort noise(13) and the path is in sendrecv. If sendrecv is absent it is assumed.

Answer

```
v=0
o=- 1563872593 1604407193 IN IP4 192.168.1.2
s=session
c=IN IP4 192.168.1.2
b=CT:1000
t=0 0
m=audio 7050 RTP/AVP 0 101 13
a=rtcp:7051
a=label:Audio
a=sendrecv
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:13 CN/8000
```

Data Collection:

At this point you have observed an issue and have collected syslogs.
You have already installed Syslog and INI viewer. In addition, you have Wireshark and its plugins installed.

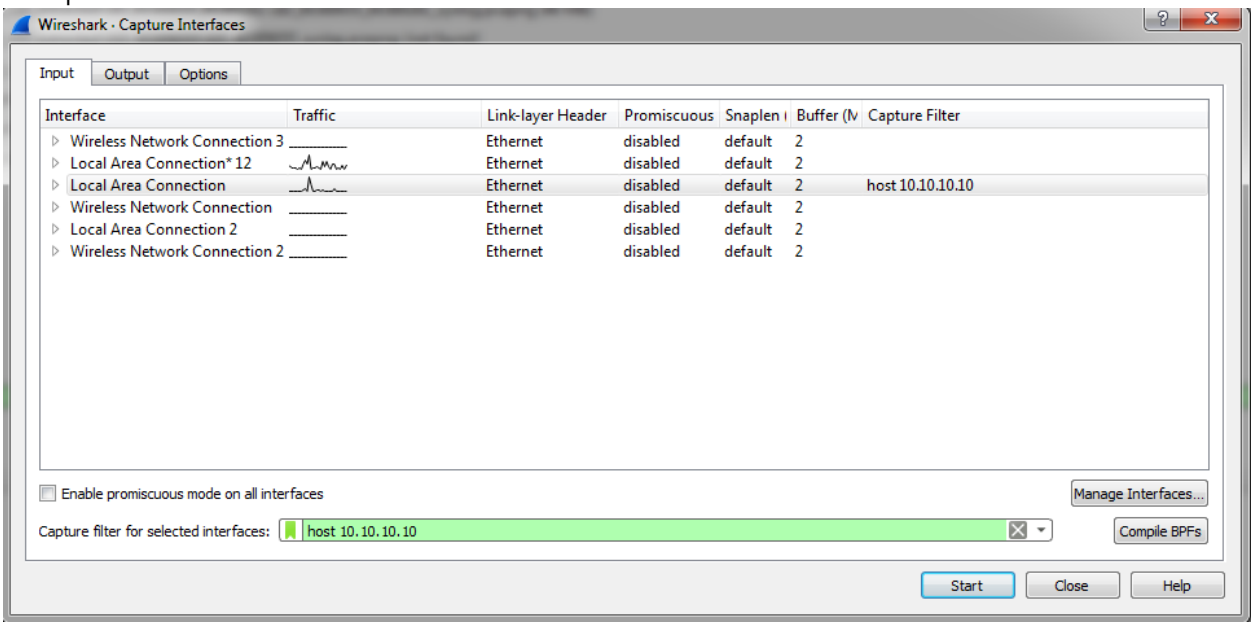
The imbedded Logging tool also known as Debug Recording will be used to capture the data. You will also be running syslog concurrently.
Ports 514, 925 and 926 are open in your network between the PC or Server running Wireshark, and the Audiocodes device. The device will send AudioCodes Debug Recording(ACDR) packets and syslog for capture.

There is an entire section in the User's Manual for setting up the Logging. It is best to isolate the call as detailed as possible during the collection. Using ANY is a last resort.

Note: The Logging tool duplicates the SIP and RTP traffic, on your network. This can degrade the network during peak traffic.

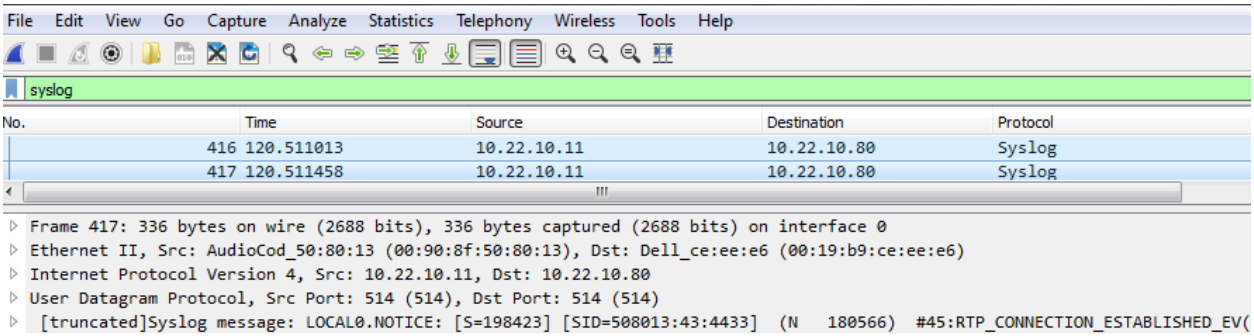
Wireshark Tips:

When setting up the Wireshark it is best to isolate the device to eliminate other traffic form being captured.
Under Capture Drop Down Menu select Options, the Wireshark Capture Interface box will appear.
Only define the interface that you will capture packets from and enter the IP addr of the Audiocodes device. Then select Start and then Stop from the main menu unit you are ready to capture your call.
The file will grow very fast due fact that RTP is sent ever 20 msec.
Example host 10.10.10.10

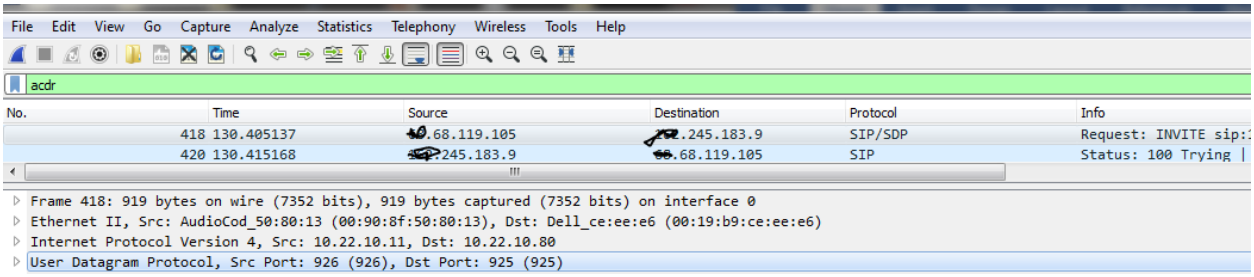


You are ready to capture data:
Start Wireshark and observer that you are collecting Syslog as well as ACDR and RTP.
You can check this by simply using the Display filter (syslog, acdr, rtp)

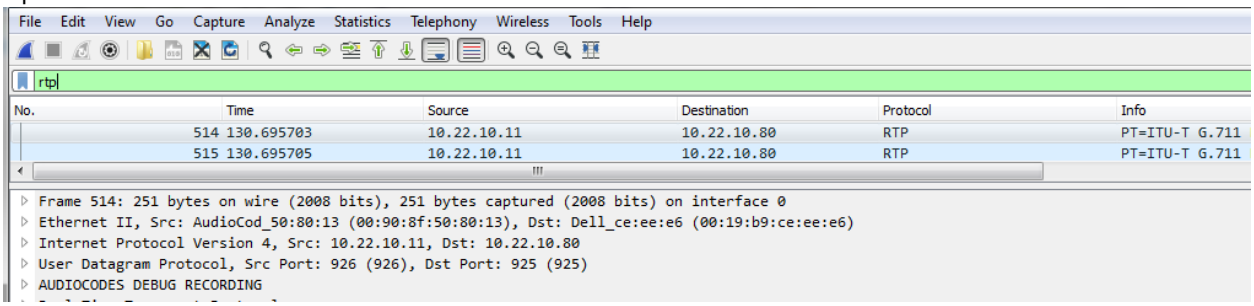
syslog



acdr(deocded shows SIP in this example)



rtp



Once you know that the packets are being captured, stop Wireshark.
Get ready to make you call, start Wireshark, make your call. Make sure the call has competed and is disconnected, then stop the Wireshark. This method will give you the smallest pcap.

Without the plugins, you cannot decode the ACDR packets, as we insert the ACDR header which our plugins decode. Wireshark will only be able to parse the data to the User Datagram. The next section will show as Data not Audiocodes Debug Recording as shown below:

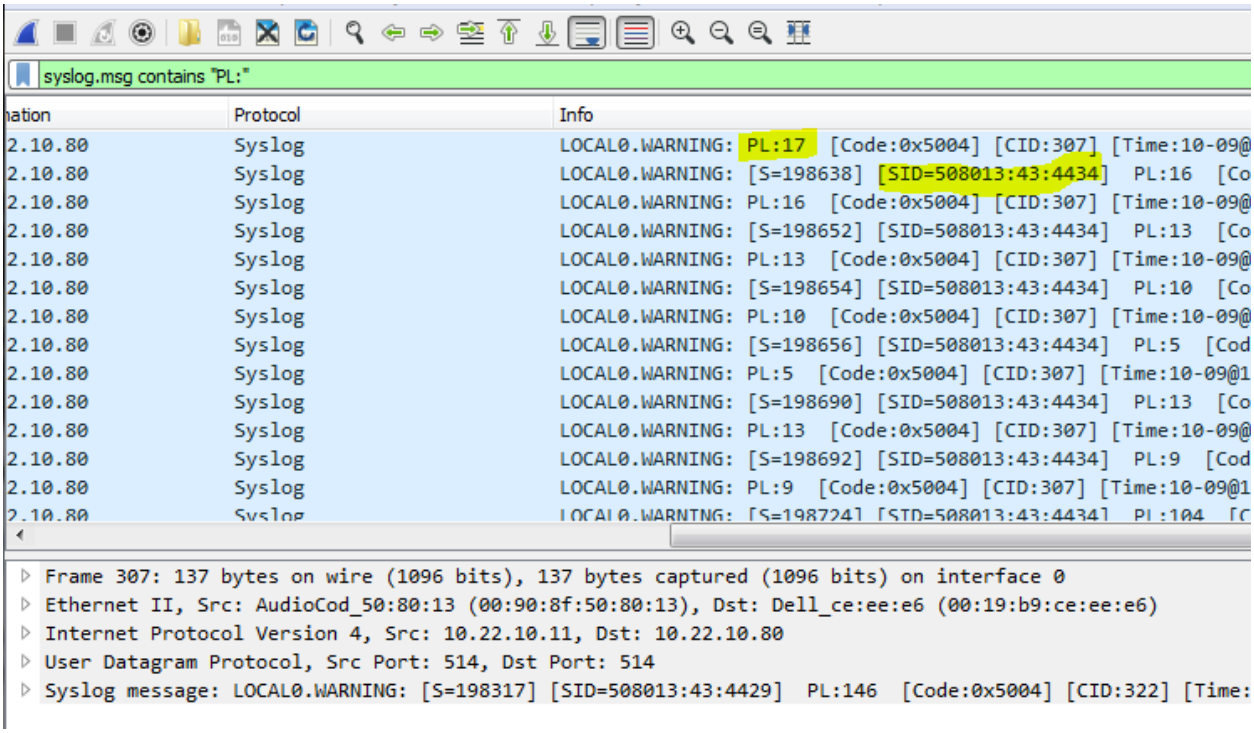
```
Frame 514: 251 bytes on wire (2008 bits), 251 bytes captured (200
Ethernet II, Src: AudioCod_50:80:13 (00:90:8f:50:80:13), Dst: Del
Internet Protocol Version 4, Src: 10.22.10.11, Dst: 10.22.10.80
User Datagram Protocol, Src Port: 926 (926), Dst Port: 925 (925)
AUDIOCODES DEBUG RECORDING
```

This is an example of an RTP packet:
The Trace Point defines the stream type,
This example is Network -> DSP…which represents from the customer's network to the Audiocodes devices DSP.
The Header Extension is shows what IP and port the RTP came from and what port on the device is was received on.
RTP Header, contains the RTP Payload, Sequence Number, and the SSRC.
SSRC is the RTP streams name/identifier.
Sequence is the number of the packet in the SSRC stream.

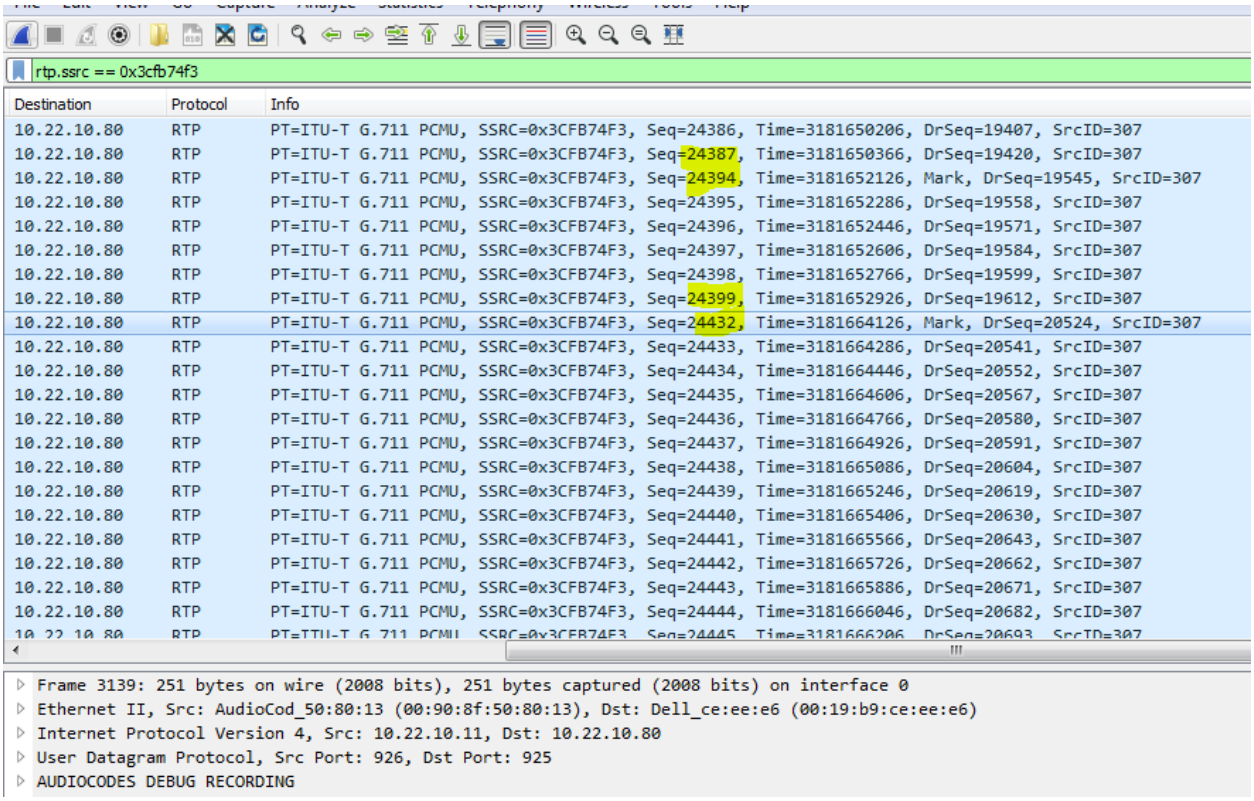
```
No.     Time          Source
    514 130.695703    10.22.10.11
    515 130.695705    10.22.10.11

Frame 514: 251 bytes on wire (2008 bits), 251 bytes captured (2
Ethernet II, Src: AudioCod_50:80:13 (00:90:8f:50:80:13), Dst: D
Internet Protocol Version 4, Src: 10.22.10.11, Dst: 10.22.10.80
User Datagram Protocol, Src Port: 926 (926), Dst Port: 925 (925
AUDIOCODES DEBUG RECORDING
    Version: 0x09
    Time Stamp: 088bd8fd (143.382781 sec)
    Sequence Number: 5129
    Source ID: 307
    Dest ID: 307
    Extra Data: 0x00
    Trace Point: Network -> Dsp (0)
    Media Type: RTP Packet (1)
    Header Extension Len: 9
  Full Session ID: 508013:43:4434
  Header Extension
      Packet source IP address: [redacted]68.119.108
      Packet source UDP port: 9386
      Packet destination UDP port: 7050
      IP type of service: 184
Real-Time Transport Protocol
    10.. .... = Version: RFC 1889 Version (2)
    ..0. .... = Padding: False
    ...0 .... = Extension: False
    .... 0000 = Contributing source identifiers count: 0
    1... .... = Marker: True
    Payload type: ITU-T G.711 PCMU (0)
    Sequence number: 23462
    Timestamp: 3181496446
    Synchronization Source identifier: 0x3cfb74f3 (1023112435)
    RTP Data (160 bytes, offset 91)
```

This is an example of Packet Loss:

When searching the syslog in the pcap use the filer syslog.msg contains "PL:"



The packets are not sequential.
Sequence number 24387 next was 24394, later 24399 skips forward to 24432.
These packets were not received by the SBC and will not be sent out of the SBC to the terminating endpoint.



rtp.ssrc == 0x3cfb74f3

Destination	Protocol	Info
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24386, Time=3181650206, DrSeq=19407, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24387, Time=3181650366, DrSeq=19420, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24394, Time=3181652126, Mark, DrSeq=19545, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24395, Time=3181652286, DrSeq=19558, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24396, Time=3181652446, DrSeq=19571, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24397, Time=3181652606, DrSeq=19584, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24398, Time=3181652766, DrSeq=19599, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24399, Time=3181652926, DrSeq=19612, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24432, Time=3181664126, Mark, DrSeq=20524, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24433, Time=3181664286, DrSeq=20541, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24434, Time=3181664446, DrSeq=20552, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24435, Time=3181664606, DrSeq=20567, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24436, Time=3181664766, DrSeq=20580, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24437, Time=3181664926, DrSeq=20591, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24438, Time=3181665086, DrSeq=20604, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24439, Time=3181665246, DrSeq=20619, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24440, Time=3181665406, DrSeq=20630, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24441, Time=3181665566, DrSeq=20643, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24442, Time=3181665726, DrSeq=20662, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24443, Time=3181665886, DrSeq=20671, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24444, Time=3181666046, DrSeq=20682, SrcID=307
10.22.10.80	RTP	PT=ITU-T G.711 PCMU, SSRC=0x3CFB74F3, Seq=24445, Time=3181666206, DrSeq=20693, SrcID=307

```
Frame 3139: 251 bytes on wire (2008 bits), 251 bytes captured (2008 bits) on interface 0
Ethernet II, Src: AudioCod_50:80:13 (00:90:8f:50:80:13), Dst: Dell_ce:ee:e6 (00:19:b9:ce:ee:e6)
Internet Protocol Version 4, Src: 10.22.10.11, Dst: 10.22.10.80
User Datagram Protocol, Src Port: 926, Dst Port: 925
AUDIOCODES DEBUG RECORDING
```

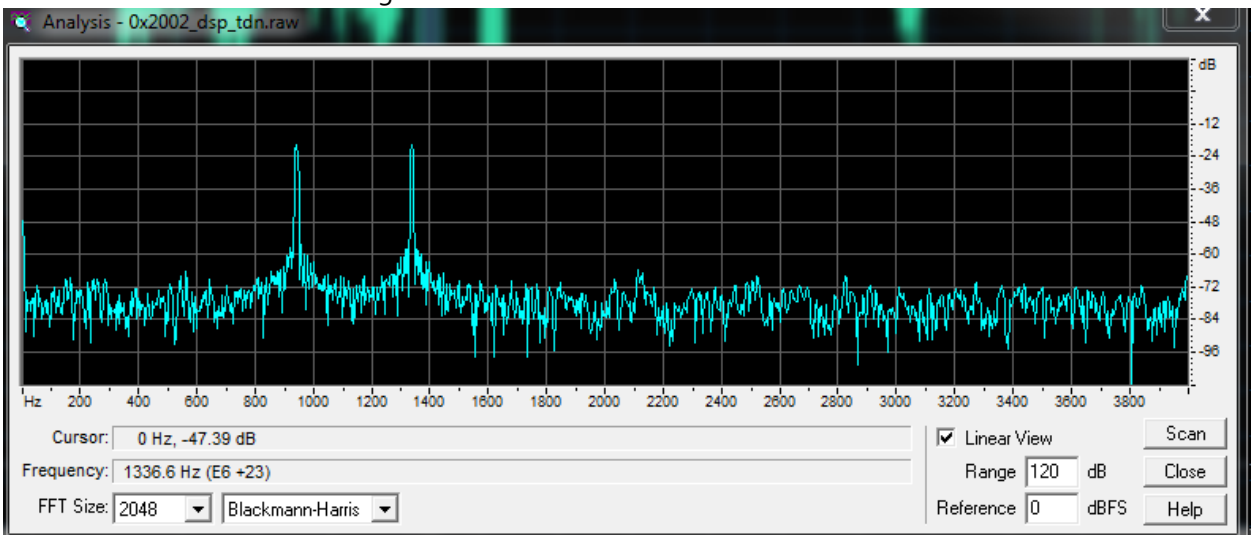
DTMF:

The following frequency chart was acquired from the DTMF wiki
https://en.wikipedia.org/wiki/Dual-tone_multi-frequency_signaling

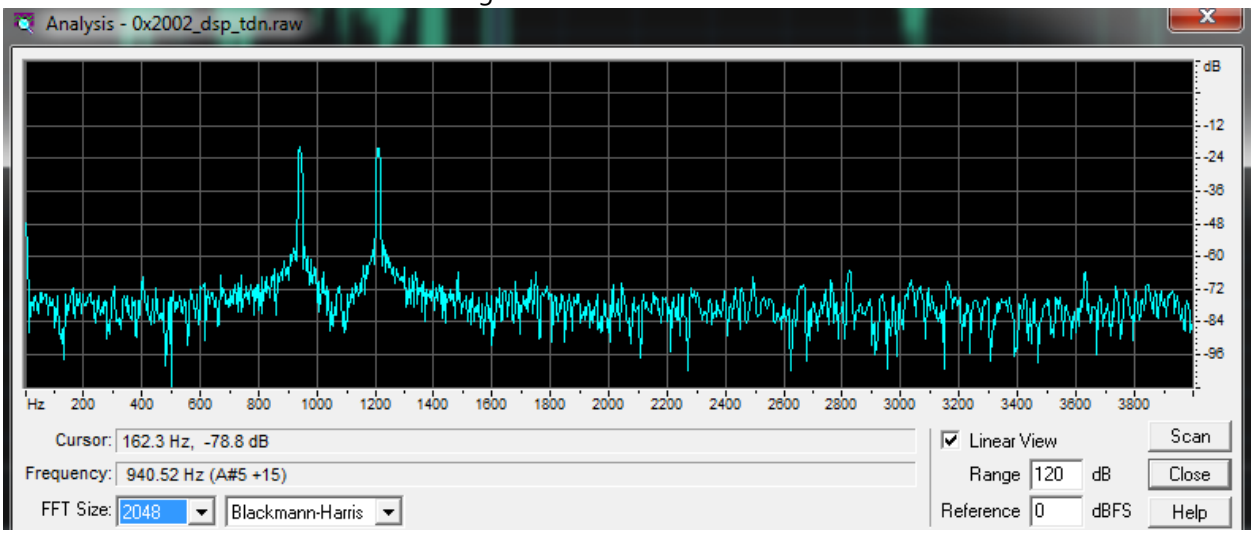
DTMF keypad frequencies (with sound clips)

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

This is 941Hz and 1336Hz the digit 0

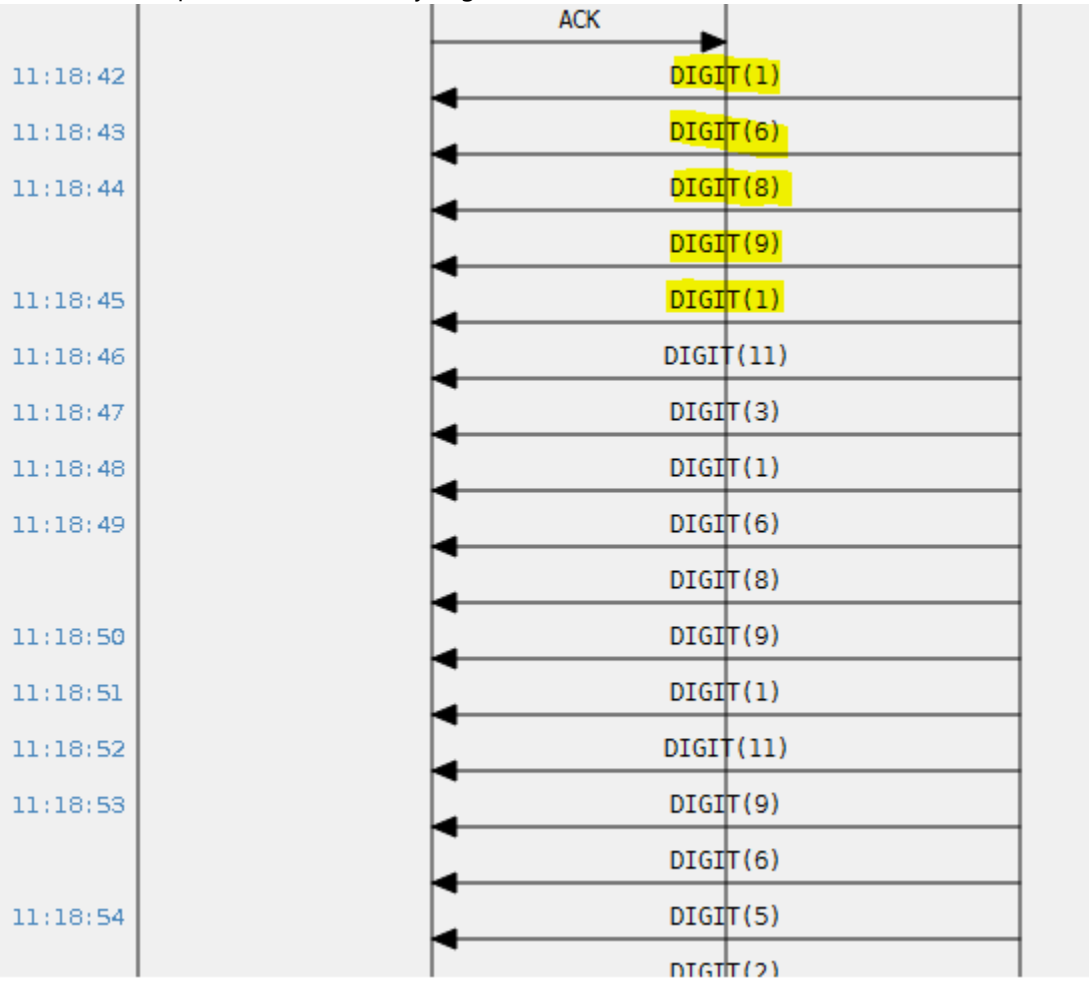


This is 941Hz and 1209Hz which is the digit *



DTMF cont.:

This is an example of DTMF in the syslog:



Here is the inband DTMF received instead of RFC2833

There is a syslog for the beginning of the digit and another syslog when the digit stops being sent. The digit 1 is for 150 msec(on time 0 and on time 150). Digit 6 is 140msec.

11:18:42.166 10.80.40.41 local0.notice [S=7636994] [SID=9f9179:58:273958] (lgr_psbrdex)(7657822) recv <-- DIGIT(1) Ch:995 OnTime:0 InterTime:32767 Direction:1 System:1

11:18:42.166 10.80.40.41 local0.notice [S=7636995] [SID=9f9179:58:273958] (lgr_flow)(7657823) (#922)ChannelResource <- (#0)NULL:DIGIT_EV

11:18:42.306 10.80.40.41 local0.notice [S=7636996] [SID=9f9179:58:273958] (lgr_psbrdex)(7657824) recv <-- DIGIT(1) Ch:995 OnTime:150 InterTime:32767 Direction:1 System:1

11:18:43.289 10.80.40.41 local0.notice [S=7637003] [SID=9f9179:58:273958] (lgr_psbrdex)(7657830) recv <-- DIGIT(6) Ch:995 OnTime:0 InterTime:970 Direction:1 System:1

11:18:43.304 10.80.40.41 local0.notice [S=7637004] [SID=9f9179:58:273958] (lgr_flow)(7657831) (#922)ChannelResource <- (#0)NULL:DIGIT_EV

11:18:43.414 10.80.40.41 local0.notice [S=7637005] [SID=9f9179:58:273958] (lgr_psbrdex)(7657832) recv <-- DIGIT(6) Ch:995 OnTime:140 InterTime:970 Direction:1 System:1

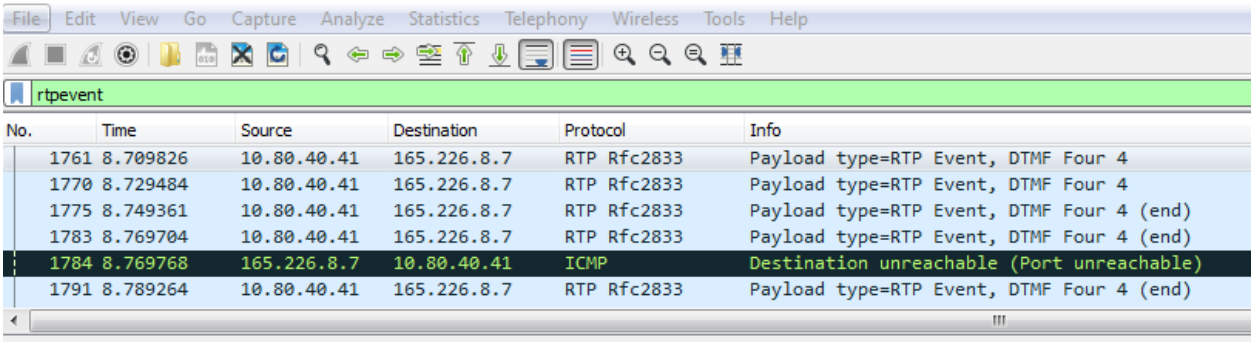
11:18:43.414 10.80.40.41 local0.notice [S=7637006] [SID=9f9179:58:273958] (lgr_flow)(7657833) (#922)ChannelResource <- (#0)NULL:DIGIT_EV

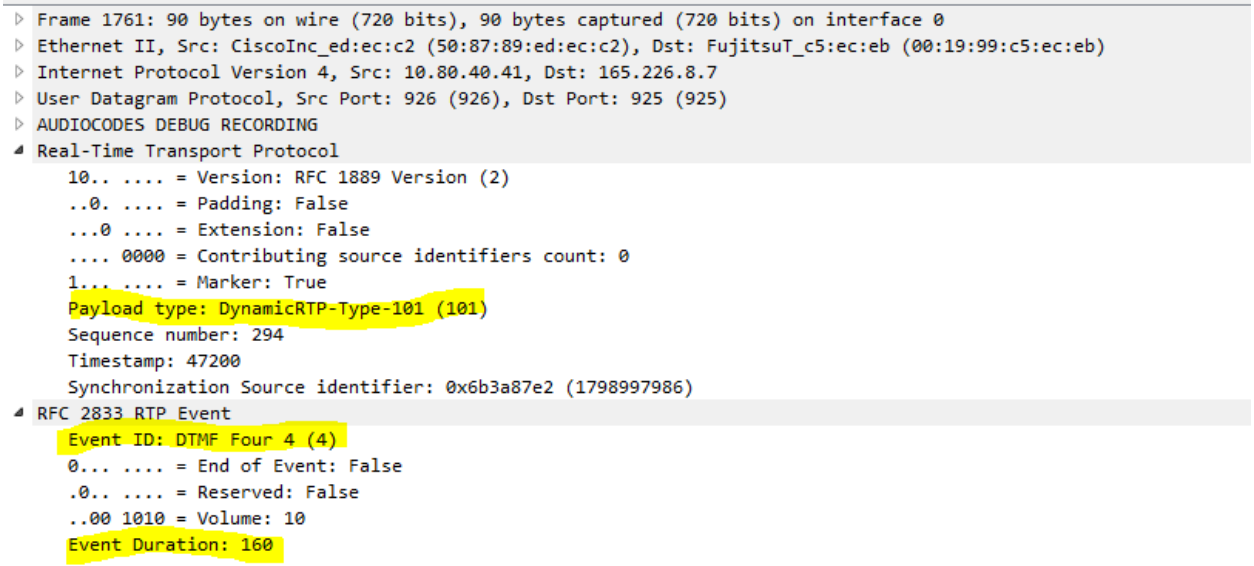
RFC2833:

This is an example of one digit, which is the digit 4.

There are multiple packets which are cumulative. Each have a duration up until the (end) of the digit.

There will be three (end) packets. This is to ensure delivery(if packet loss is present) to stop the digit form being played indefinitely.





3.5 Payload Format

The payload format is shown in Fig. 1.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|     event     |E|R| volume    |          duration             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

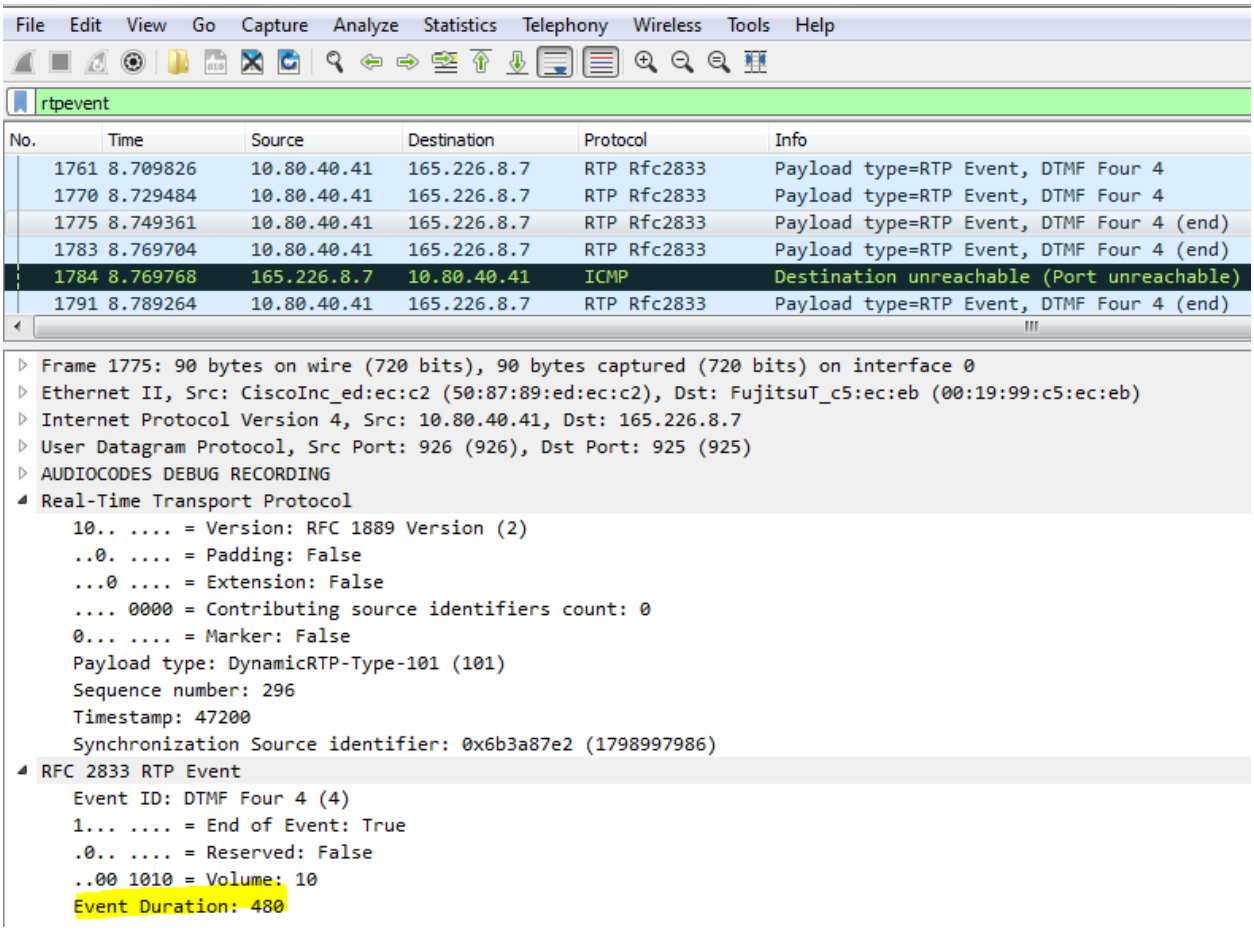
From RFC2833

volume: For DTMF digits and other events representable as tones, this field describes the power level of the tone, expressed in dBm0 after dropping the sign. Power levels range from 0 to -63 dBm0. The range of valid DTMF is from 0 to -36 dBm0 (must accept); lower than -55 dBm0 must be rejected (TR-TSY-000181, ITU-T Q.24A). Thus, larger values denote lower volume. This value is defined only for DTMF digits. For other events, it is set to zero by the sender and is ignored by the receiver.

duration: Duration of this digit, in timestamp units. Thus, the event began at the instant identified by the RTP timestamp and has so far lasted as long as indicated by this parameter. The event may or may not have ended.
For a sampling rate of 8000 Hz, this field is sufficient to express event durations of up to approximately 8 seconds.

E: If set to a value of one, the "end" bit indicates that this packet contains the end of the event. Thus, the duration parameter above measures the complete duration of the event.

The end digit with have the total duration of the packets. Again, the end digit is the same packet but is sent three times.



For any further questions regarding this topic or other technical topics:

- Contact your AudioCodes Sales Engineer
- Visit our AudioCodes Services and support page at https://www.audiocodes.com/services-support
- Access our technical documentation library at https://www.audiocodes.com/library/technical-documents
- Access to AudioCodes Management Utilities is available at https://services.audiocodes.com/app/answers/detail/a_id/20
- Contact Technical Support to submit a support ticket at https://services.audiocodes.com