



# Troubleshooting Common Issues in VoIP

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# Voice over Internet Protocol (VoIP)

## Introduction

Voice over IP, or VoIP, refers to the delivery of voice and multimedia over IP networks. The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream.<sup>1</sup>

Since the transmission of the packet occurs through an IP network, the quality of communications is inherently less reliable than a circuit switched public telephone network. There is no network mechanism to ensure that data packets are not lost and are delivered in sequential order. Essentially, it is a best effort without Quality of Service (QoS) guarantees.

Several metrics are available for measuring VoIP call quality. This paper will provide a basic understanding of VoIP traffic and of the quality metrics used to monitor VoIP calls.

## VoIP Metrics

Maintaining high quality VoIP calls can be difficult as VoIP is more sensitive to network delays and packet loss compared to any other network applications. VoIP quality is measured based on the following metrics:

- Network Jitter and Delay
- Packet data Loss
- Latency
- Mean Opinion Score (MOS)

### Network Jitter and Delay

Real-time voice communications over the network are sensitive to delay in packet arrival time or packets arriving out of sequence. Excess jitter results in calls breaking up. Jitter can be reduced to a certain extent by using jitter buffers. Jitter buffers are small buffers that cache packets and provide them to the receiver in sequence and evenly spaced for proper playback. Buffer lengths can be modified, however, if jitter buffer is increased too much then the call will experience an unacceptable delay. Consequently, a reduction in buffer turns results in less delay but more packet loss. Jitter is measured in milliseconds (ms).

### Packet data Loss

Packet loss occurs when one or more packets of data fail to reach their destination. A single packet loss is referred to as “packet gap”, and series of packet loss is known as” burst”. Packet loss can occur for a variety of reasons including link failure, high congestion levels, misrouted packets, buffer overflows and a number of other factors. Packet loss causes interrupted playback and degradation in voice quality. Packet loss can be controlled using packet loss concealment techniques within the playback codec.

### Latency

Latency, or lag, is the time delay caused in the transmission of a voice packet. Excess latency results in delay and finally to echo. Latency is measured in milliseconds (ms)

### Mean Opinion Score (MOS)

Mean Opinion Score is a numerical value to indicate the perceived quality of the call from the user’s perspective of the received call after compression, transmission, and decompression. MOS is a calculation based on the performance of the IP network and is defined in the ITU-T PESQ P.862 standard and is expressed as a single number in the range of 1 to 5, where 1 is lowest perceived quality and 5 is the highest.

Desirability Scale	R-Factor	MOS Value
Desirable	94 - 80	4.4 - 4.0
Acceptable	80 - 70	4.0 - 3.6
Reach Connection	70 - 50	3.6 - 2.6
Not Recommended	50 - 0	2.6 - 0

### R-Factor

The R-factor is a score that takes into account both user perceptions (User R-factor) and the effect of equipment impairments (Network R-factor) to arrive at a numeric expression of voice quality. These metrics are calculated by a formula that balances all equipment impairments and perception factors and is reported as a

single number on a per-call basis, typically in the range of 15 to 94. The MOS and R-Factor helps in determining the QoS. The R-factor is calculated using a basic formula called E-Model as shown below.

$$R\text{-Factor} = Ro - Is - Id - Ie + A$$

*R-Factor: Overall network quality rating (ranges between 0 and 100)*

*Ro: Signal to noise ratio*

*Is: Impairments simultaneous to voice signal transmission*

*Id: Impairments delayed after voice signal transmission*

*Ie: Effects of Equipment (MOS is converted to Ie)*

*A: Advantage factor (attempts to account for caller expectations)*

## Call Detail Records (CDRs)

A CDR is a data record that documents the details of a phone call and fall into one of two classifications: the basic CDR, and the diagnostic CDR or CMR (Call Monitor Records). The basic CDR includes information about the call such as originating phone number, receiving phone number, call starting time, call duration, billing details, and other call details. The CMR includes metric information needed for monitoring VoIP calls including addresses, status (successful, failed, transferred), number of packets, packet bytes, packet loss, start time, initial setup duration, duration, current jitter, maximum jitter, MOS, R-factor, QoS for each call, number of packets that arrive out of order, detailed analysis for packet loss and delay, gap and burst measurements, and long-term Call Detail Records trending.

## Monitoring VoIP Performance

In order to properly troubleshoot VoIP call quality, you should have a number of tools at your disposal that monitor and measure the critical call quality components. VoIP monitoring typically falls into one of three categories:

- Network monitoring tools that monitor VoIP performance based on network performance statistics
- Protocol analyzers
- Dedicated VoIP tools
- Synthetic VoIP traffic generators

## Network Monitoring Tools

Network monitoring tools are non-intrusive, or passive, and examine each stream of voice traffic across the network to estimate the MOS score and calculate the R-Factor. The use of network monitoring tools has the advantage that all calls in the network can be monitored without any additional network overhead. For the IT professional, monitoring VoIP using network monitoring tools is essentially just another network task and, therefore, typically the most practical approach.

### **Protocol Analyzers**

Protocol analyzers are hardware or software tools that capture and analyze VoIP traffic packets and calculate jitter, and latency directly from the packet stream.

### **Dedicated VoIP Tools**

Most dedicated VoIP tools were originally developed for the telecomm industry and are great for testing IP phone and gateway designs but not as good at solving deployment problems within the network.

### **Synthetic VoIP Traffic Generators**

Synthetic VoIP traffic generators reduce time to the readiness of VoIP services, while assuring they meet quality requirements as perceived by users.

## **Troubleshooting VoIP Quality Issues**

Quality VoIP calls require a quality IP network that can deliver voice packets within the minimum requirements around jitter, packet loss, and latency. Network monitoring systems and specifically network traffic monitoring systems can help the IT professional better understand the impact of network performance on VoIP call quality. At a minimum, you need to analyze at the end points (both call origination point and call destination point), however, measuring at additional points within the call path will help further narrow the potential problem areas.

**Step 1:** Check the error/status code to indicated failed calls—see error description. (In case of a fail call go to step 4)

**Step 2:** Check MOS score. If it's bad, try to find calls placed at the same time and verify if they MOS was similar

**Step 3:** Analyze packet loss, latency or jitter in order to see what could be a potential problem (slow network, unreliable network, overloaded routers or switches)

**Step 4:** If possible do comparison of the calls that were placed between the same locations at the same time to confirm symptoms

**Step 5:** Look for the related gateways and call managers and their performance status (usually via call connection topology)

**Step 6:** Try to see call path and all related network element like routers and switches and see their performance metrics.

**Step 7:** Verify channel utilization at the time of the call.

**Step 8:** Verify call related end-point devices (soft IP phones, HW phones)

**Step 9:** Establish VoIP synthetic operation between the same locations to prevent the same issue and continuously monitor situation

## Best Practices in VoIP

- Benchmark your network health before and after your deployment
- Automate monitoring and measurement of the call quality components and metrics
- Measure at multiple points to see exactly where call quality drops
- Upgrade your equipment to support QoS, PoE, layer 3 protocols, and multiple VLANs
- Determine the proper CODEC based on your WAN and LAN capabilities
- Monitor your gateways and channel utilization
- Monitor your call path network devices performance – mainly routers and switches.

## References

<sup>1</sup> Voice over IP. (2012, June 6). In Wikipedia, The Free Encyclopedia. Retrieved 15:35, June 12, 2012, from [http://en.wikipedia.org/w/index.php?title=Voice\\_over\\_IP&oldid=4962430](http://en.wikipedia.org/w/index.php?title=Voice_over_IP&oldid=4962430)

## Top 5 reasons to Download SolarWinds VoIP & Network Quality Manager (VNQM)

1. Analyze Call Detail Records generated by Cisco® Call Manager & Avaya® Communication Manager
2. Automatically determine and deploy IP SLA
3. Be notified when key VoIP metrics such as jitter, latency, packet loss, or MOS exceeds thresholds
4. Monitor and troubleshoot VoIP call performance for Cisco & Avaya
5. Gain detailed visibility into performance of Cisco VoIP gateway and PRI trunk utilization

Get a global snapshot of your IP SLA operations with SolarWinds VNQM!

 **DOWNLOAD FREE TRIAL**



**VoIP & Network Quality Manager**

**All IP SLA Operations** (MANAGE OPERATIONS | HELP)

GROUPED BY OPERATION TYPE, OPERATION

- HTTP
- DHCP
- DNS
- FTP
- ICMP Echo
- TCP Connect
- UDP Echo
- VoIP UDP Jitter
- ICMP Path Jitter

**VoIP CallManagers** (HELP)

HOST	PHONES		GATEWAYS/ENDPOINTS		REJECTED LAST POLL
	UNREGISTERED	REJECTED	UNREGISTERED	REJECTED	
AUS-SUB-01	4%	1%	0%	39%	11/18/2013 4:31:09 AM
AUS-PUB-01	100%	0%	0%	0%	11/18/2013 4:30:53 AM

**Cisco IP SLA Operations Map** (HELP)

WAN QUALITY METRICS FROM REMOTE PERFORMANCE MONITORS

Map showing global IP SLA operations with labels: Dns - Ottawa, Bastogne, Core, thxSalesForce.com, UdpEcho BAA > Core, tcp - Bas SMTP.

**Calls by Region** (HELP)

LAST 15 MINUTES

Region	Call Count
AUS (AUS-SUB-01)	42 call(s)
LAF (AUS-SUB-01)	22 call(s)
NewZealand (AUS-SUB-01)	20 call(s)
IRL (AUS-SUB-01)	14 call(s)
SIN (AUS-SUB-01)	13 call(s)
Rest	20 call(s)
<b>TOTAL:</b>	<b>131 call(s)</b>

**IP SLA Operations Health Overview** (HELP)

Health Status	Count
Operation Up	130
Operation Warning	1
Operation Critical	2
Operation Down	2
Unknown	2
Operation Unreachable	0

**Search VoIP Calls** (HELP)

Call Origination:

**Operations with Issues** (HELP)

## About SolarWinds

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