

Voice over IP Networks

Voice over IP sometimes gets a bad press because the quality is often poor. This is not a fault of VoIP itself, but is a result of the way that VoIP uses a network. A VoIP network is far more unforgiving of errors – especially temporary errors.

Errors

VoIP uses UDP – a protocol that is normally only seen in DNS lookups, and DNS applications always retry if a lookup fails. Most other traffic on a network is TCP/IP which was specifically designed to withstand packet loss and to perform retries.

VoIP uses Real Time Protocol and it sends a UDP packet containing 20ms of speech all the time that a VoIP connection is up. Running alongside RTP is Real Time Control Protocol (RTCP) that can be used to determine the quality of the connection. If the packets are deemed not to be getting through then the call is closed.

This makes VoIP particularly vulnerable to transient network errors such as those associated with poor quality network cabling. A network cable under too much strain may suffer from occasional disconnects resulting in packet loss. Such errors are of course masked by any TCP/IP connection as that protocol explicitly allows for errors, meaning that the faulty cable just appears slow if indeed any fault is seen with it.

The CEO of a company got a lot of dropped calls every afternoon. The problem was eventually tracked down to faulty wiring in the Ethernet socket in his office, which expanded due to the heat in the office and made on and off connections. He hadn't noticed that his web surfing went slower in the afternoon.

There's also less leeway in people's minds about the experience they expect about phones as opposed to computers. Everyone is familiar with having to restart your computer, but they are not used to having to restart their phones. They are used – perfectly understandably – to picking up the phone and always getting a dial tone. They don't expect their calls to be cut off.

It is therefore important that the network cabling is checked thoroughly before installing a VoIP system. If you don't then the user ends up with a poor experience and you get the blame.

Quality of service in the office

Most modern office networks run at 100M or 1000M. Although a modern computer is pushed to deliver data up the theoretical 1G maximum if the network is running at 100M then this bandwidth can be all used up by a simple file copy. The use of switches as opposed to hubs has helped reduce bottlenecks, but there are still a large number of hubs about.

As described above, VoIP traffic expects to get to its destination. It also expects to get there as quickly as possible and for the packets to arrive in the correct order. VoIP technology commonly includes what is known as a "jitter buffer" which allows for packets arriving in the

wrong order, but there is a trade-off between the size of the buffer and the increased latency that a large buffer brings with it. A latency of more than 300ms means that VoIP calls sound very poor.

It therefore a basic premise that when installing a VoIP system you need to keep the voice traffic away from the data traffic. Often this can be achieved by simply having two LANs, one for the phones and one for the data. They can be two physical LANs on distinct routers or they can be Virtual LANs on the same router. Often a physical separate router that can provide power over internet is often useful.

The problem comes when phones that have an extra socket on the back are used. These are intended to support a user plugging in a laptop and only needing one internet connection on the wall. In this case the laptop will be sending data over the phone network and a download onto the laptop and network speed will cause problems if they are also making a call.

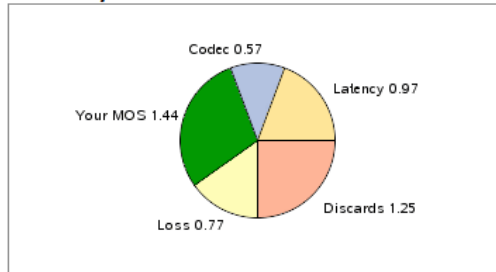
The solution is to use VLANs. All such phones are VLAN aware and can send Ethernet packets that are tagged with the VLAN they want to use. Laptops don't send VLAN tags so are allocated into VLAN 1 by the switch.

Quality of service out of the office

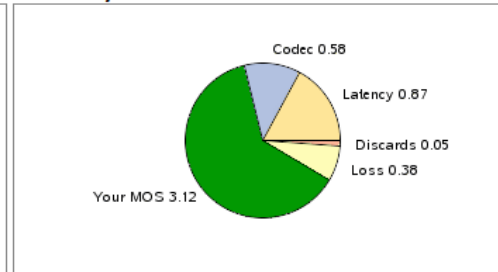
The problems described above about traffic requirements with the local area network are more problematic when using the Internet. In general you have little control over how your VoIP traffic is treated by your ISP and the ISP of your destination. That is why we still recommend using normal landlines for the majority of the phone traffic.

But there are of course times when you want to use VoIP over the Internet. Probably the biggest problem is your own ADSL line. The following figure shows the performance of a VoIP call while a broadband upload and download is taking place.

MOS Analysis From You TO London



MOS Analysis FROM London To You



Media Quality

MOS	<ul style="list-style-type: none"> ■ 1.4 / 5.0 (Best with G.711 is 4.4) 								
Degradation Sources	<table border="1"> <tbody> <tr> <td>■ Codec</td> <td>0.57 16.1%</td> </tr> <tr> <td>■ Latency</td> <td>0.97 27.1%</td> </tr> <tr> <td>■ Packet Discards</td> <td>1.25 35.1%</td> </tr> <tr> <td>■ Packet Loss</td> <td>0.77 21.6%</td> </tr> </tbody> </table>	■ Codec	0.57 16.1%	■ Latency	0.97 27.1%	■ Packet Discards	1.25 35.1%	■ Packet Loss	0.77 21.6%
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■ Packet Discards	1.25 35.1%								
■ Packet Loss	0.77 21.6%								
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)								
Round-Trip Latency	681 ms								
Packet Discards	6.3%								
Packet Loss	3.9%								
Loss Periods	Min: 20 ms Avg: 140 ms Max: 460 ms <i>Burst Loss</i>								
Jitter	Min: 0 ms Avg: 15 ms Max: 770 ms								

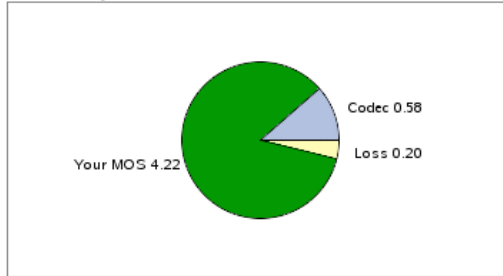
Media Quality

MOS	<ul style="list-style-type: none"> ■ 3.1 / 5.0 (Best with G.711 is 4.4) 								
Degradation Sources	<table border="1"> <tbody> <tr> <td>■ Codec</td> <td>0.58 30.6%</td> </tr> <tr> <td>■ Latency</td> <td>0.87 46.2%</td> </tr> <tr> <td>■ Packet Discards</td> <td>0.05 2.9%</td> </tr> <tr> <td>■ Packet Loss</td> <td>0.38 20.2%</td> </tr> </tbody> </table>	■ Codec	0.58 30.6%	■ Latency	0.87 46.2%	■ Packet Discards	0.05 2.9%	■ Packet Loss	0.38 20.2%
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Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)								
Round-Trip Latency	681 ms								
Packet Discards	0.1%								
Packet Loss	0.9%								
Loss Periods	Min: 20 ms Avg: 20 ms Max: 60 ms <i>Random Loss</i>								
Jitter	Min: 2 ms Avg: 6 ms Max: 43 ms								

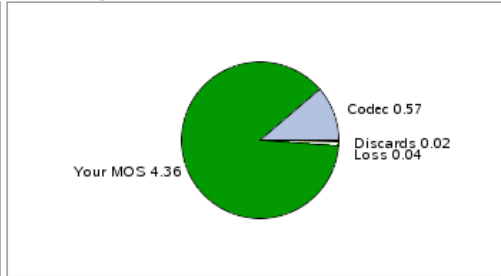
The actual figures are unimportant but you can see that there's a latency of over 680ms, over 6% of the traffic is discarded and there is almost half a second of missed traffic. This would not have been a good experience if it was real call rather than a simulation.

The situation can be easily resolved by using traffic shaping. Sometimes this can be done by turning on Quality of Service (QoS) at the ADSL modem, but often this is not supported. In this case it's possible to run all the local network traffic through a single machine that is the only device connected to the ADSL modem. So long as you can tell that machine the upload speed you consistently get then the machine can avoid sending too much data to the modem so that the modem's buffers never fill. The machine can then prioritise the VoIP traffic before sending it onwards. The figure below shows the effect when the same test is run.

MOS Analysis From You TO London



MOS Analysis FROM London To You



Media Quality

MOS	<p>■ 4.2 / 5.0 (Best with G.711 is 4.4)</p> <p>Degradation Sources</p> <table border="1"> <tr> <td>Codec</td> <td>0.58</td> <td>73.9%</td> </tr> <tr> <td>Latency</td> <td>0.00</td> <td>0.0%</td> </tr> <tr> <td>Packet Discards</td> <td>0.00</td> <td>0.0%</td> </tr> <tr> <td>Packet Loss</td> <td>0.20</td> <td>26.1%</td> </tr> </table>	Codec	0.58	73.9%	Latency	0.00	0.0%	Packet Discards	0.00	0.0%	Packet Loss	0.20	26.1%
Codec	0.58	73.9%											
Latency	0.00	0.0%											
Packet Discards	0.00	0.0%											
Packet Loss	0.20	26.1%											
Codec	G.711 (PCM at 64kbps, 20ms RTP payload, 80kbps IP BW)												
Round-Trip Latency	128 ms												
Packet Discards	0.0%												
Packet Loss	0.8%												
Loss Periods	Min: 20 ms Avg: 20 ms Max: 20 ms <i>Random Loss</i>												
Jitter	Min: 0 ms Avg: 5 ms Max: 18 ms												

Media Quality

MOS	<p>■ 4.4 / 5.0 (Best with G.711 is 4.4)</p> <p>Degradation Sources</p> <table border="1"> <tr> <td>Codec</td> <td>0.57</td> <td>89.7%</td> </tr> <tr> <td>Latency</td> <td>0.00</td> <td>0.0%</td> </tr> <tr> <td>Packet Discards</td> <td>0.02</td> <td>3.3%</td> </tr> <tr> <td>Packet Loss</td> <td>0.04</td> <td>6.9%</td> </tr> </table>	Codec	0.57	89.7%	Latency	0.00	0.0%	Packet Discards	0.02	3.3%	Packet Loss	0.04	6.9%
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Jitter	Min: 4 ms Avg: 7 ms Max: 27 ms												