Jitter

Jitter is a common problem of the connectionless networks or packet switched networks. Because the information (voice packets) is divided into packets, each packet can travel by a different path from the sender to the receiver. When packets arrive at their intended destination in a different order than they were originally sent, the result is a call with poor or scrambled audio.

Jitter is technically the measure of the variability over time of the latency across a network, and it is one of the most common VoIP call quality problems.

Steps to get rid of Jitter:

1. ***Check Network Infrastructure***: Verify all devices physical connections

   - Router
   - Switches
   - IP Phones
   - Computers

2. ***Enable QoS***

Quality of service (QoS) is the overall performance of a telephony or computer network, particularly the performance seen by the users of the network. To quantitatively measure quality of service, several related aspects of the network service are often considered, such as error rates, bit rate, throughput, transmission delay, availability, jitter, etc. Quality of service is particularly important for the transport of traffic with special requirements. In particular, much technology has been developed to allow computer networks to become as useful as telephone networks for audio conversations, as well as supporting new applications with even stricter service demands.

3. If you're router does not provide QoS capabilities, ***set up "Port Forwarding"***

Port forwarding opens certain ports on your home or small business network, usually blocked from access by your router, to the Internet. Opening specific ports can allow games, servers, BitTorrent clients, Voip providers and other applications to work through the usual security of your router that otherwise does not permit connections to these ports.

Set up two forwarding entries the "Port Forwarding" (or similar) configuration form on the NAT configuration interface, each of which cause the NAT device to forward all traffic destined for the designated range of port numbers to the fixed IP address of the SIP phone:

- SIP signaling: Ports **5060 to 5099**
- RTP audio: Ports **10000 to 20000**
- Remove SIP Transformation
- Service: Any
- Allow incoming and outgoing traffic from these IP Address Ranges:
  Source: WAN, Address Range 208.73.1.120 to 208.73.1.130
  Source: WAN, Address Range 208.73.2.120 to 208.73.2.130
- Destination: LAN, Address Range * to *
- Comment: ActivePBX Servers

- If there is an option for 'Allow Fragmented Packets, then please enable this options:
  TCP** Connection Inactivity Timeout (minutes) to **60
  UDP** Connection Inactivity Timeout (seconds) to **3600

4. ***Upgrade internet bandwidth:***

It's time to upgrade your office's Internet bandwidth if you notice any of these signs:

- Web pages' load slowly
- Downloads don't fully complete because your connection is saturated. The server thinks you no longer want or need the file or you've gone offline so your request is cancelled
- If you don't host your own email, mailboxes do not finish synchronizing or the email program reports errors because the initial sync can't complete before it attempts to receive new messages
- Web forms submitted aren't sent successfully, instead a time out or error message is received or the page never comes back

5. ***You may restart all your network equipment (router/modem, switches) and it will get rid of packet loss and temporarily fix the issue*.**
   - Unplug all equipment for 1 minute, and connect the router/modem first, once all lights are on, then plug in your switches. This is also automatically restart all your desk phones.

6. ***If you continue to have audio issues and/or high Jitter results, contact your Internet Service Provider to help you find a solution.***