Factsheet

Tests – VOIP

VOIP

<u>Voice over IP (UDP)</u>

Supported clients: Whiteboxes, Routers

The Voice over IP (VoIP) test measures the quality of a voice call between the client and a nearby test server. It is intended to characterise how suitable the user's broadband connection is for placing and receiving VoIP calls.

The test uses the SIP protocol for signalling and can support a variety of codecs for the call itself. By default, the test uses the G.711 codec, which uses a bi-directional stream of 64kbps UDP traffic. Other codecs are supported, including G.722. The duration of the test is configurable, but will run for 10 seconds by default.

The test may use a SamKnows-provided SIP server for signalling, or may alternatively be configured to use a third-party SIP server. This allows for third-party SIP services to be tested on an end-to-end basis, assessing the entire VoIP infrastructure, not just the quality of the end user broadband network.

The test captures the following metrics:

- Round trip latency
- Downstream packet loss
- Upstream packet loss
- Downstream jitter
- Upstream jitter
- MOS (Mean Opinion Score)