Testing Technologies

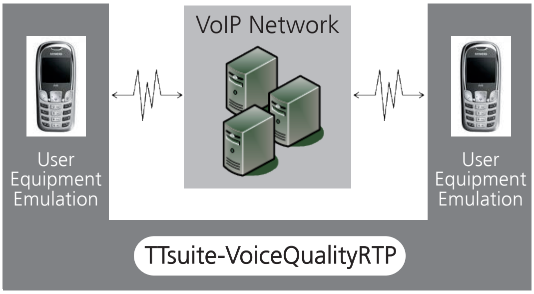
TTsuite-VoiceQualityRTP

The Test Suite for Testing Voice Quality in VoIP Networks

TTsuite-VoiceQualityRTP is a powerful test suite for speech quality measurements. It provides various functions to give information about the quality of a SIP based VoIP connection such as the MOS value or the delay, and many other parameters.

Features and Benefits

- 42 test cases to measure voice quality (PESQ, E-Model)
- SIP protocol to build up a connection
- Speech analyzing (MOS) according to
  - E-Model (ITU-T Rec. G. 107)
  - PESQ (ITU-T Rec. P.862)
- Gathering RTP session parameters
  - RTPPackets and RTCPPackets
  - Bytes, PacketLoss and SSRC
  - deltaTMean, TMax and deltaTMin
- Testing the voice quality of SIP signaled RTP stream
  - User agents / clients
  - Networks
- Fully integrated into TTworkbench
- Full test automation
- Easy trouble shooting



Speech Analysis for VoIP Network

Supported Audio Codecs

- PCMU
- PCMA
- GSM
- G.723
- G.726/32
- G.729
- iLBC

Reference Platforms

- Java 6
- Microsoft Windows XP, Vista, 7 (x86-32)

TTsuite

 • More information at www.testingtech.com