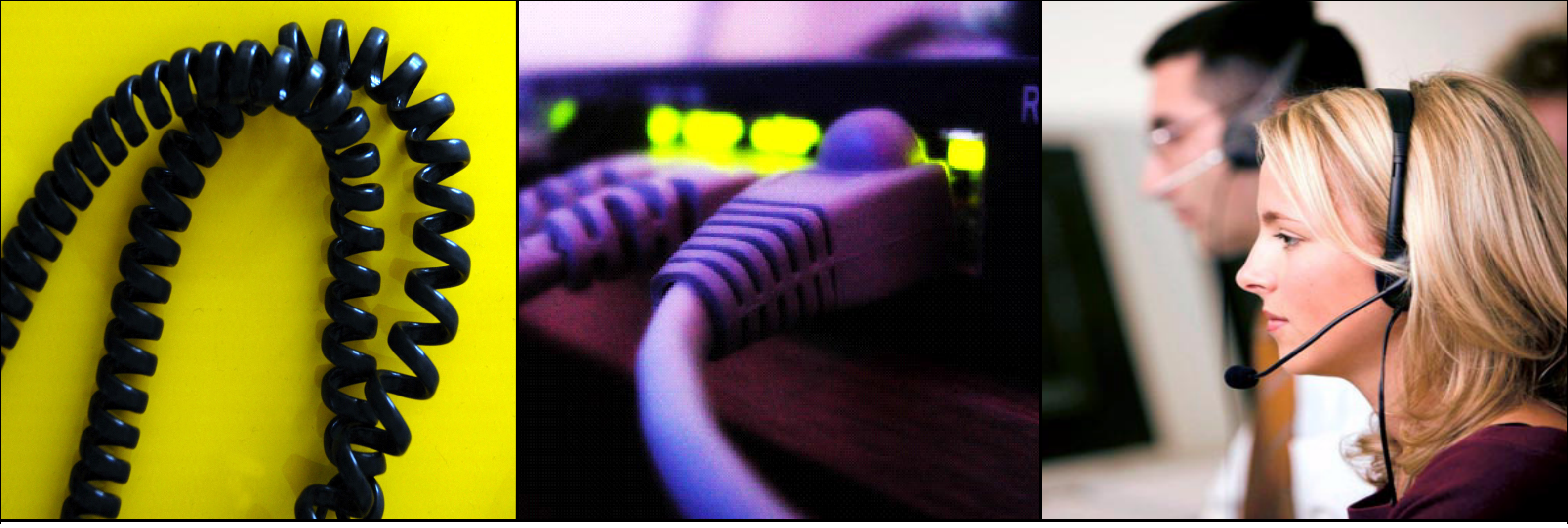


VoIP.Master (service turn-up)

in Test we Trust

VoIP.Master delivers installation, maintenance and service assurance tools for VoIP and Unified Communications. Designed to meet the needs of VoIP technicians. With a powerful feature set VoIP.Master provides a comprehensive test capability required for next generation voice environments.

Based on a USB memory device, VoIP.Master is a self-contained live software environment that can be run on most x86/X64 based laptops or computers.

By using a live USB memory device users are able to turn existing laptop or computer assets into a powerful VoIP tester without the concern of anti-virus or other corporate lock-down issues that may be present when using the device in a native Windows mode.

Key Features and Benefits

VoIP.Master offers the most comprehensive support for VoIP turn-up and maintenance testing through its unique functional testing approach. By providing this capability VoIP.Master can emulate key VoIP/UC infrastructure elements allowing users to quickly test and ensure the correct operation and performance of VoIP networks and equipment.

IP PBX and Network Emulation

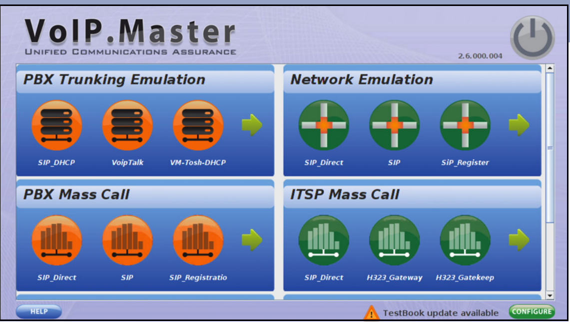
Users can for example connect VoIP.Master to SIP trunks and VoIP networks emulating an IPBX, making multiple VoIP calls ensuring the trunk is operational and performing to pre-agreed Service Level Agreements.

“Users may connect VoIP.Master to SIP trunks to emulate an IPBX in order to verify the SLA and the network and VoIP user devices”

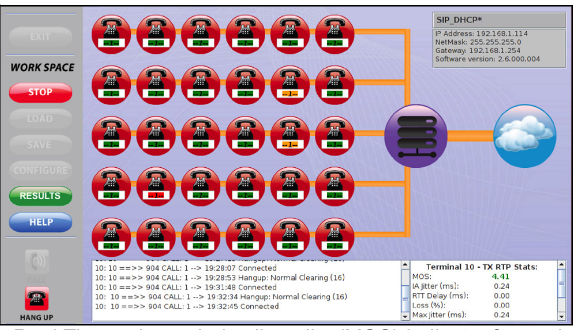
VoIP.Master can also emulate a SIP VoIP network allowing VoIP equipment to be tested without the need for an operational SIP trunk or network, ideal to verify VSIP VoIP Network and SIP VoIP Equipment the prior to deployment by means of emulation modes as standard. VoIP.Master also has the ability to emulate up to five T.38 fax machines then users can use PDFs to send or receive fax.

Test reports can be customised to include the logo and the name and address of customer’s circuit/service being tested.

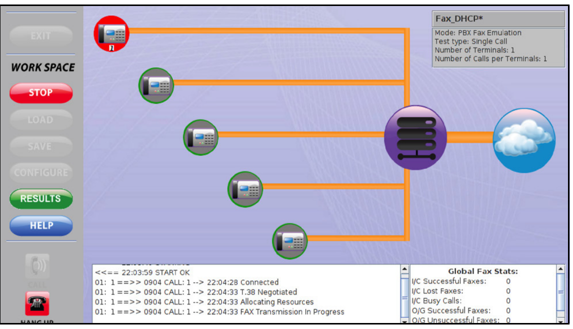
ALBEDO Telecom



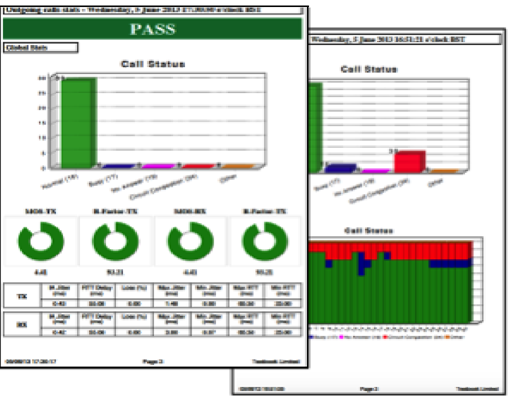
Simple Graphical User Interface with user-defined profiles provides users with one button setup capability.



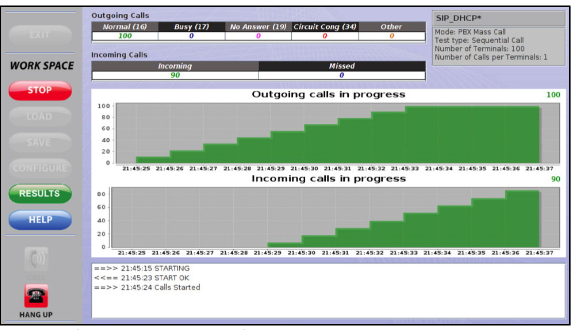
Real Time color coded call quality (MOS) indicator for each call with detailed media metrics saved to PDF test report.



T.38 Fax emulation allows the VoIP network to be tested for its ability to successfully transmit and receive Fax over IP.



PDF reports provide an audit trail of tests performed.



Mass Call mode allows SIP trunks and VoIP devices to be tested for call capacity under outgoing and incoming calls.

IPBX **assured** installation

Powerful emulation

VoIP.Master can support up to 30 simultaneous VoIP calls using 'Virtual Terminals' in PBX and Network emulation mode. Any combination of outgoing/incoming calls is supported, incoming calls on answer are presented with an auto-attendent capability providing users the ability to select the required mode of operation for that call. The workspace provides a single graphical view of the test in process with call statistics, test log and call status all visible. Users can configure all test parameters within the workspace and save a test profile for future recall though a single button.

- Single, Sequential, Bulk Call modes.
- Call quality metrics.
- E-Model (ITU-T G.107)support.
- Outgoing and incoming call support with ability to generate hundreds of calls in bulk call mode.
- Option of sending WAV file, looping or connecting to internal microphone and speaker on call connection.
- Multiple codec support.

Mass Call mode

With the increase in availability and speed of WAN services, the call capacity of SIP trunks and VoIP equipment is increasing. For VoIP installations in larger enterprises there is sometimes the need to test calls in excess of the 30 call capacity of the standard PBX and ITSP emulation modes. Mass Call mode provides a quick and easy way of testing call capacity of SIP trunks and VoIP equipment, supporting up to 200 simultaneous calls with or without media.

- Emulation of up to 200 calls (in/out).
- Equipment and Network emulation.
- Reports with call quality metrics.

Call Quality metrics

Ensuring call quality is paramount to ensure VoIP services meet customer expectations and that service providers reduce customer churn. VoIP.Master provides a GUI real-time call quality (MOS) indicator for each call in progress whilst also saving call metrics to a PDF test report in the background. The real-time quality indicator allows users to view the performance of SIP Trunks and VoIP circuits as the number of calls in progress are increased.

- Real-time coloured coded MOS for each terminal (call).
- Detailed media (RTP) statistics for each call.
- Pass/Fail thresholds for MOS
- Pass/Fail thresholds for RTP metrics (Jitter/Delay/Loss).

VoIP.Server

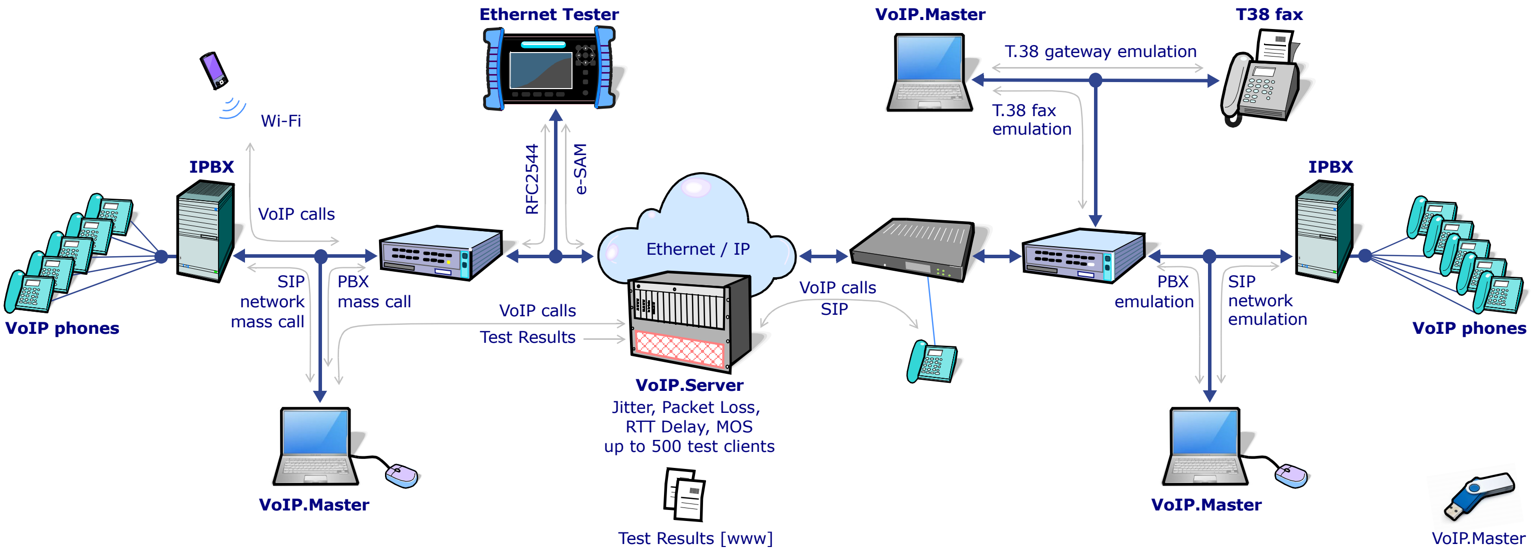
This is centralized version which is based on a Server that can work with VoIP.Master or directly with VoIP terminals to improve the test process and increase technician efficiency and productivity:

- Provides a repeatable and consistent test environment.
- Facilitates advanced VoIP tests to improve testing and troubleshooting.
- Provides a central database to store test results from VoIP.Master.
- View test results through a Web browser using built-in Web Server.
- Test Customers equipment and network access for correct operation and configuration using VoIP.Server.

Testing Process

ProcessVoIP.Master can make test calls to the VoIP.Server which terminates incoming calls, supporting different test types such as Loop, WAV File (RTP), CLI Announcement and Call-Back. On completion of the test, VoIP.Master sends its test results to the VoIP.Server and are stored in the database.
VoIP.Server can also be used to test Customer VoIP Equipment and Network Access prior to service turn-up. Calls are made to the VoIP.Server to test call connectivity and call quality with test results stored on the server. Test results are accessible through the VoIP.Server Web interface.

Remote customer self serve testing is possible using VoIP.Server and a downloadable windows test client. Customers download and run the windows test client which will register with VoIP.Server which will then automatically run through a series of tests. Test results are stored on the database and can be viewed through the web server interface using a standard browser.



Spot the difference

Automatic software update

The optional software support package provides users with an online update capability. When connected to the Internet VoIP.Master will check the VoIP.Master application server and, if a new version is detected, will alert the user giving the option to download and update VoIP.Master automatically.

In the rapidly changing VoIP and Unified Communications environment the software support package provides users with protection for their investment by ensuring that VoIP.Master always has the latest features.

KEY FEATURES

- PBX Trunking Emulation
- SIP Trunking Emulation
- PBX Mass Call Mode
- SIP Mass Call Mode
- T.38 Fax Emulation
- T.38 Gateway Emulation
- MOS indication results
- Detailed RTP statistics
- Pass/Fail thresholds
- Call Log and Test Statistics
- Change Log
- Automatic Notification
- Emulate VoIP infrastructure
- Automatic Software Update
- Centralized Reports

Testing with VoIP.Master

Some samples of VoIP troubleshooting.

- *Trunks Dropping:* test SIP trunk for stability under load through number of simultaneous calls and bulk call.
- *Poor Call Quality:* check audio paths and call quality (MOS) with different IP quality of service settings.
- *Codec Mismatch:* test call connection capability using different Codecs.
- *Call Completion:* check registration capability and repeatability.
- *Trunk Registration Failure:* test ability to make calls to different destinations (Mobiles, fixed, ...)
- *One Way Audio/No Audio:* check paths on all calls (bi-directional) on SIP trunk

T.38 fax emulation

When enterprises install a VoIP telephony infrastructure they often move their existing Fax services to Fax over IP also known as ITU T.38. It is therefore important to be able to check that the VoIP network and T.38 gateway operate correctly and that successful fax transmission can take place.

BENEFITS

- Efficient provision SIP trunks and PBX VoIP equipment
- Comprehensive VoIP tests in three clicks
- Quality test (30 simultaneous calls) in one minute
- Best price/functionality call capacity of SIP trunks
- Improves efficiency

VoIP.Master has the ability to emulate up to five T.38 fax machines then users can use PDFs to send or receive fax.

- T.38 fax and SIP network emulation
- Send / Receive fax or in-band (G.711)
- Received fax as PDF and save them
- Send PDF as fax

Test reports

Comprehensive PDF test reports are produced automatically in the background each time you run a test ensuring you have a record of tests performed. The reports can be customised to include the logo on the top of the page and may also include additional information such as the name and address of customer's circuit/service being tested.

Two types of reports are produced: *Call Log* provides a record of all the calls that VoIP.Master makes showing individual call status, while *Test Statistics* provides a graphical report of call completion and call quality assessment for each terminal.

- Graphical test results.
- Cutomisable reports with logo.
- Detailed Call Log and Statistics reports.
- Test Pass/Fail indication with thresholds.

USERS

- VoIP operators
- VoIP PBX Installers/
- VoIP/IPBX suppliers
- Large enterprise companies

Operation	
Call modes	• VoIP.Master: Single Call, Sequential Call, Mass Call
PBX Emulation	• Emulation of an IPBX or VoIP device on a SIP trunk • Up to 30 simultaneous outgoing/incoming calls • Media quality measurement (E-Model MOS) for each call and trunk • Simple one button tests through stored user profiles • Ability to call different numbers (destinations), number ranges, etc • Mass Call mode up to 200 simultaneous outgoing/incoming calls to test trunk capacity
SIP Network Emulation	• Emulation of a SIP Trunk/VoIP Network • Up to 30 simultaneous outgoing/incoming calls • Media quality measurement (E-Model MOS) for each call and trunk • Simple one button tests through stored user profiles. • Ability to call different numbers (destinations), number ranges, etc • Mass Call mode up to 200 simultaneous outgoing/incoming calls to test PBX
T.38 Fax Emulation	• Emulation T.38 Fax machines at the customer premise allowing test faxes to be sent or received • Up to five simultaneous outgoing/incoming Fax calls • SIP network/trunk test for successful Fax over IP (T.38) operation • Support for both T.38 and G.711 Fax pass-through modes • Received Fax's automatically converted to PDF for local viewing • Comprehensive PDF test reports with customisable header/logo
T.38 Gateway Emulation	• Emulation of a T.38 Fax Gateway/Network environment • Up to five simultaneous T.38 fax devices to be tested both receiving and sending test faxes of VoIP.Master
Reports	• Comprehensive PDF test reports with customisable header/logo

Feature	VoIP.Master-Nano	VoIP.Master-Lite	VoIP.Master	VoIP.Server
PBX emulation	yes	yes	yes	n.a.
SIP Network emulation	no	yes	yes	yes
T.38 Fax emulation	no	no	optional	n.a.
Num of Terminals	1	5	30	30
Simultaneous Calls	no	no	200	up to 500
G.711, G.722, G.726, GSM, iLBC, Speex	no	yes	yes	yes
G.729 Codec	no	no	yes	yes
E-model (MOS-R factor)	yes	yes	yes	yes
Perceptual Voice Quality Testing	no	no	no	yes
Calls to PSTN	yes	yes	yes	yes
SIP, RTP, UDP, TCP, TLS, SRTP	yes	yes	yes	yes
DTMF tone	yes	yes	yes	yes
SIP registration	yes	yes	yes	yes
VLAN	yes	yes	yes	yes
TOS/COS	yes	yes	yes	yes
PDF reports	yes	yes	yes	yes
Online software update	optional	optional	optional	optional
Test results storage	local storage	local storage	local storage	central database
NAT Support	no	no	no	Stun, ICE, TURN
ARP and Trace Route	yes	yes	yes	n.a.
Call Modes	Single	Single, Sequential	Single, Sequential, Mass	Single, Loop
WAV file, CLI Ann., Call-Back	no	no	no	yes

Ordering Information	
AT-VMST-N	Provides PBX (CPE) emulation modes supporting 1 Terminal/Call. Includes MOS measurement, RTP statistics, DTMF tone and reports.
AT-VMST-L	Provides PBX (CPE) and ITSP (Network) emulation modes supporting up to 5 Terminals and simultaneous Calls in Single, Sequential and Bulk call modes. Includes MOS measurement, RTP statistics, DTMF tone and pdf reports
AT-VMST	Provides PBX (CPE) and ITSP (Network) emulation modes supporting up to 30 Terminals and simultaneous Calls in Single, Sequential and Bulk call modes. It is supplied in a USB stick and a Single Floating License to be executed on a user PC or labtop. Includes MOS measurements, RTP statistics, DTMF tone pdf reports and one G.729 codec license and Mass Call mode supporting 200 simultaneous calls in PBX and ITSP to test trunk capacity.
AT-VMST-L-G729	G.729 Codec. Adds G.729 codec capability for outgoing calls.
AT-VMST-T38	T.38 Fax. Adds T.38 Fax emulation capability to the Standard and Plus versions.
AT-TBS-000012	Annual Software Maintenance - 12 Months. Provides Software maintenance License for VoIP.Master allowing on-line upgrade of all new software releases including new feature releases for the licensed software options only.

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