# **VPP**

### Detect Reproduce Analyse

VoIP network behaviour in test networks and live networks

### What is it?

Malden VoxPort Packet is a reference VoIP end-point designed for performance validation, troubleshooting, analysis and ongoing reporting. It delivers vital and timely information about test and production versions of fixed and mobile (VoLTE) networks. Product Development, Quality Assurance and Tier 3/4 support personnel will appreciate the ease of deployment and flexibility of VoxPort Packet.

### How does it work?

VoxPort Packet (VPP) generates real voice calls over a VoIP network, analysing voice quality and other parameters affecting users' perception of performance. Optional built-in packet impairment generation and managed codec rate changes combine to make VoxPort Packet+ a simple and effective way to understand, manage and even reproduce VoIP network behaviour.

	VPP	VPP+	VPPf	VPP+f
Typical applications	Basic, fixed license	Advanced, fixed license	Basic, floating license	Advanced, floating license
Active Monitoring – network-wide				
VoIP network QA or lab test				
VoLTE test – live network				
VoLTE test with base station emulator				
Mobile modem test				

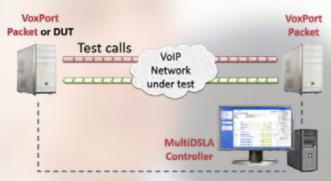




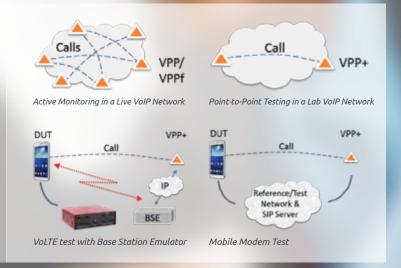


### How is it used?

Test calls can be made with or without SIP server registration. The VPP encodes and transmits real speech samples. It also receives and decodes the speech. A test system controller, such as MultiDSLA, manages the configuration, runs the test call plan and stores the measurements for reporting and analysis. VPP is highly flexible and can be configured in many ways to address specific test requirements. With VoxPort Packet you have the tools to examine, understand and even reproduce VoIP network behaviour.

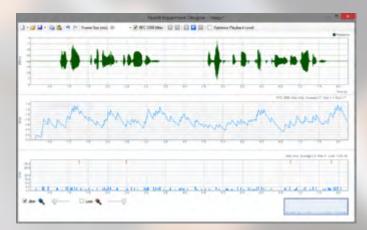


VoxPort Packet runs on a Windows platform. VPP+ offers enhanced features.

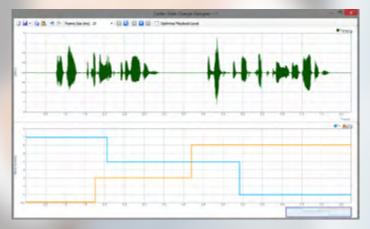


# Overlays: The key to reproducing and understanding speech quality impairment VPP+ has two overlay modes: packet impairment and rate-change.

The Packet Impairment Overlay defines a precise pattern of packet impairments. The Rate Change Overlay defines transmit rate changes or Codec Mode Requests (CMR) synchronised to speech transmissions. These overlays may be constructed by the user with a graphical overlay designer. Alternatively, the Packet Impairment Overlay may be based on a capture from a live network. The example here shows a user-defined jitter/loss overlay, which is synchronised to the start of a test event to ensure repeatability.



This Impairment Overlay defines a repeatable pattern of loss and jitter changes



This Rate Change Overlay defines a repeatable pattern TX rate and Codec Mode Request changes

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## Excellent test efficiency and economy in one VoIP test solution!

As well as enabling advanced VoIP testing, VPP+ combines with MultiDSLA Controller to provide synchronised packet impairment and codec rate change events on a single platform – replacing and out-performing a test system, impairment generator and automation host:



A Test System with VoxPort Packet+ eliminates the need for automation and impairment platforms

# Why synchronise packet impairments to the speech signal?

When an RTP packet is lost in a VoIP network the effect may be noticeable, or may be completely unnoticeable. Exploring the effects of packet impairment during high-energy or low-energy speech, and during silence, is important in analysing the performance of error concealment techniques. It follows that to perform repeatable tests of the effects of packet impairments on voice quality, the impairments must be synchronised to the speech signal used for testing: this is exactly what VPP+ does. Randomly-generated loss and jitter have their place in testing of course, but do not allow repeatable testing.

AMR-WB with packet loss examples and corresponding effects



# Malden VoxPort Packet & VoxPort Packet+ Datasheet

Feature	VoxPort Packet	VoxPort Packet+
Compatibility MultiDSLA Controller v5.0 and later	•	•
Codec support G.711, G.729, G.729A, G.729B, G.723.1, G.722, G.726, iLBC, Opus*, AMR NB & WB with DTX, GSM EFR*, 8k, 16k, 32k linear pcm Frame sizes 5, 10, 20, 30, 40, 50, 60ms according to codec User defined static jitter buffer	All	All
Call capacity available per platform	1-5, 10, 20 or 30	1-5, 10, 20 or 30
Network test interface definable for each call	•	•
IPv4 / IPv6 support	•	•
Signalling		
SIP port definable	•	•
SIP call setup and teardown: RFC 3261	•	•
SIP over TLS*	•	•
RTP: RFC 3550	•	•
RTCP*	•	•
Secure RTP*	•	•
SDP: RFC 4566	•	•
Digest Authentication: RFC 2617	•	•
Route and Record-Route*	•	•
IP QoS, DiffServn IEEE 802.1Q VLAN tagging*	•	•
Caller ID on received call*	•	•
DTMF RFC 4733*	•	•
SIP-less – Call established to user defined IP address and ports		•
Impairment generation		Random packet loss % User defined packet loss Random jitter within range User defined jitter



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Feature	VoxPort Packet	VoxPort Packet+
AMR codec rate changes within call		Managed transmit rate change Managed Codec Mode
		Request
AMR codec modes		
(a) no negotiation	•	•
(b) negotiation within defined parameters		•
	Control	Control
Packet Capture	Signalling	Signalling
	RTP	RTP
Test signal generation		
Normal – voice file only	•	•
Complex – voice + background noise files	•	•
User-defined background noise throughout call*	•	•
White/Pink/Gaussian background noise*	•	•
RTP loopback, receive to send*	•	•
Operating System support 64 or 32 bit Windows 7 Professional, 8 Pro, 2008 and 2012 Servers	Runs as Windows Service	Runs as Windows Service
	Floating, static or	Floating, static or
Licensing	transferable	transferable
	within MultiDSLA	within MultiDSLA

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