ALBEDO VoIP.Master

unified communications assurance

ALBEDO Telecom

the Path to Excellence

About ALBEDO Telecom

ALBEDO Telecom offers a full range of telecommunication products and services to the international market.

- **Hand-held Filtering Taps**: battery operated, 1kg. double port
- **Stream-to-disk appliance**: SSD disk, wirespeed capture, wirespeed storage, 2Gb/s
- **Impairment Generator**: Carrier Ethernet and IP
- **Hand-held testers**: E1, SDH, GbE, SyncE, IP, IPTV, VoIP, Datacom, Jitter, Wander
- **Acceptance Labs**: IPTV, VoIP, ISDN, POTS
- **Consultancy / Integration**: IPTV, VoIP

SIP Market – The Opportunity

SIP trunking has become the "connectivity choice" for enterprises upgrading communications systems to IP PBX and unified communications (UC)

- The global SIP trunking services market is on track to grow 35% in 2014, to $4.4 billion
- SIP trunking will be the fastest growing business VoIP segment through 2017
- Cloud PBX and unified communication services a $12 billion market by 2018
- SIP trunks are at less than a 15 percent penetration rate, the it has plenty of room to grow

(Source: Infonetics - October 2014)

VoIP.Master – VoIP Products

VoIP.Master Lite

VoIP.Master

VoIP.Master-S

USB VoIP.Master Lite

Cost effective SIP Trunk and VoIP services for installers

USB VoIP.Master

Advanced SIP Trunk and VoIP services tester

Ideal for experts installing SIP Trunks, VoIP services, VoIP equipment and hosted VoIP services

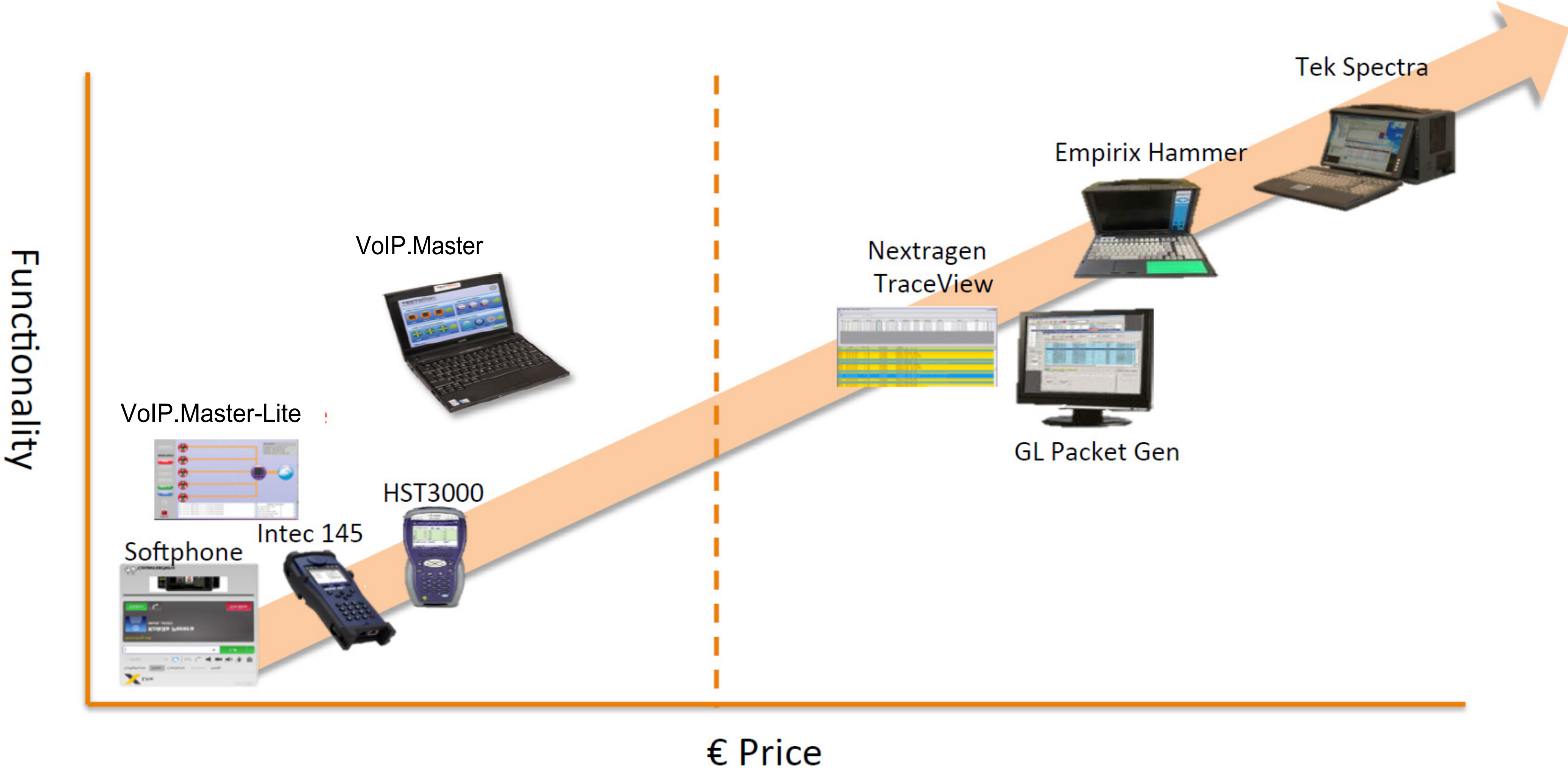
VoIP.Master-S

Centralised VoIP test capability for SIP trunk and hosted VoIP providers

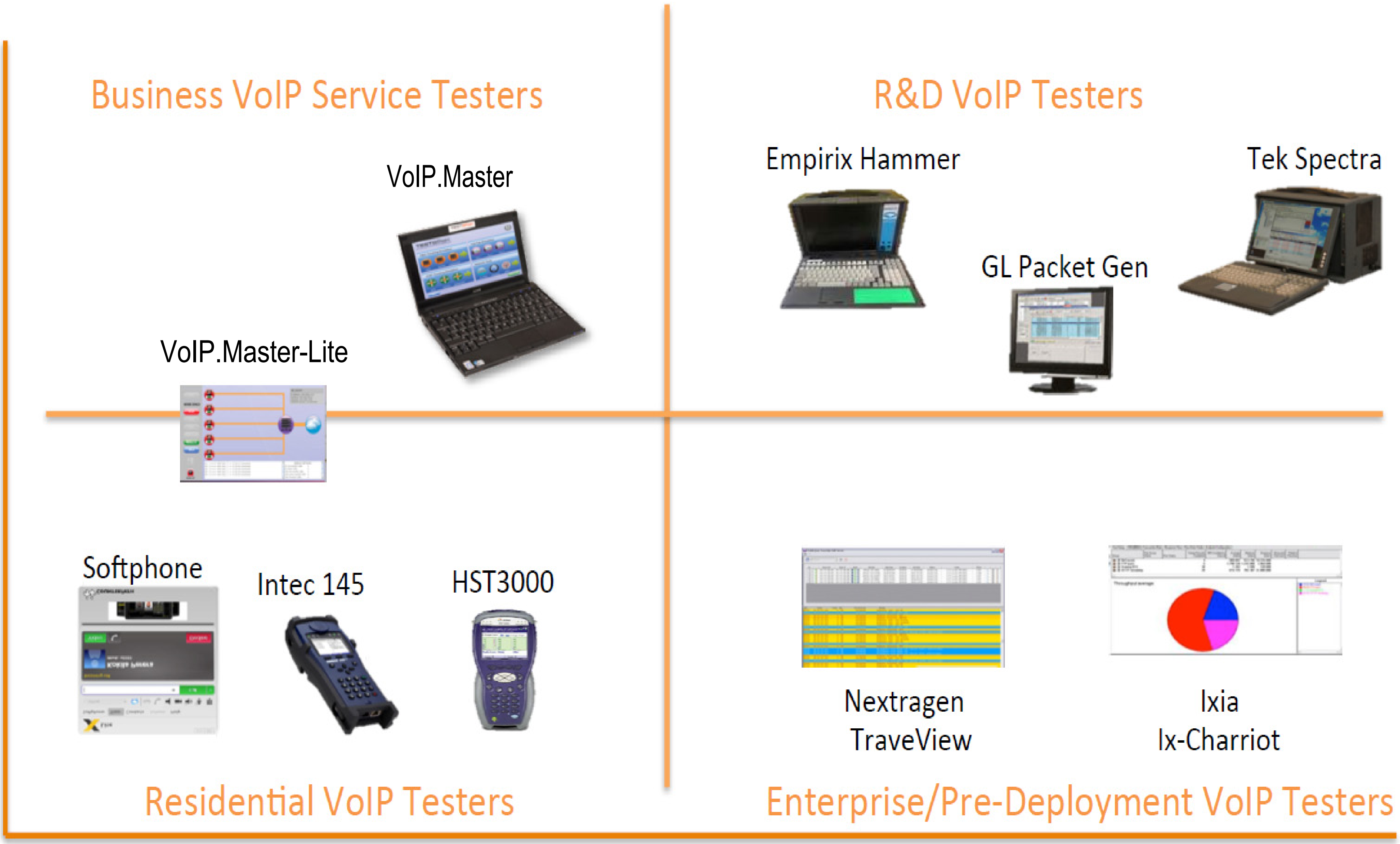
Target **Markets**

- Technicians/Engineers turning-up, installing, maintaining VoIP services and products.
  - **Tier 1 Service Providers** – BT, UPC, Vodafone etc.
  - **Tier 2 Service Providers** – Spitfire, Gamma, Vanilla IP etc.
  - **VoIP/IPBX and equipment suppliers** – Siemens, Cisco, Panasonic, NEC etc.
  - **3rd Party suppliers**, resellers and installers of VoIP equipment.
  - **Large enterprise**, public utilities, government agencies with their own VoIP network or equipment

VoIP.Master **Positioning**



VoIP Active Testing – **Market Quadrants**



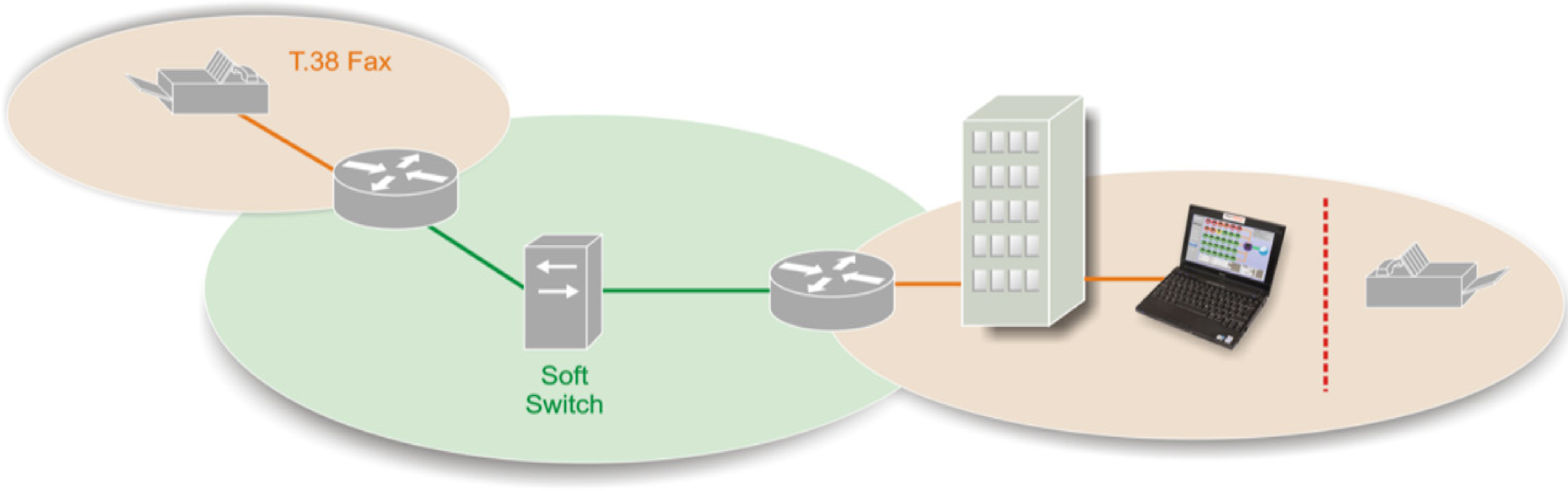
PBX emulation

- VoIP.Master can **emulate an IPBX** to checking an SIP trunk for call operation and call quality
  - 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
  - Comprehensive PDF **test reports** with customisable header/logo
  - Ability to call different numbers (destinations), number ranges etc
  - Different **call modes**: Single Call, Sequential Call, Bulk Call
  - Mass Call mode **up to 200 calls** to test trunk capacity

IP **Network** emulation

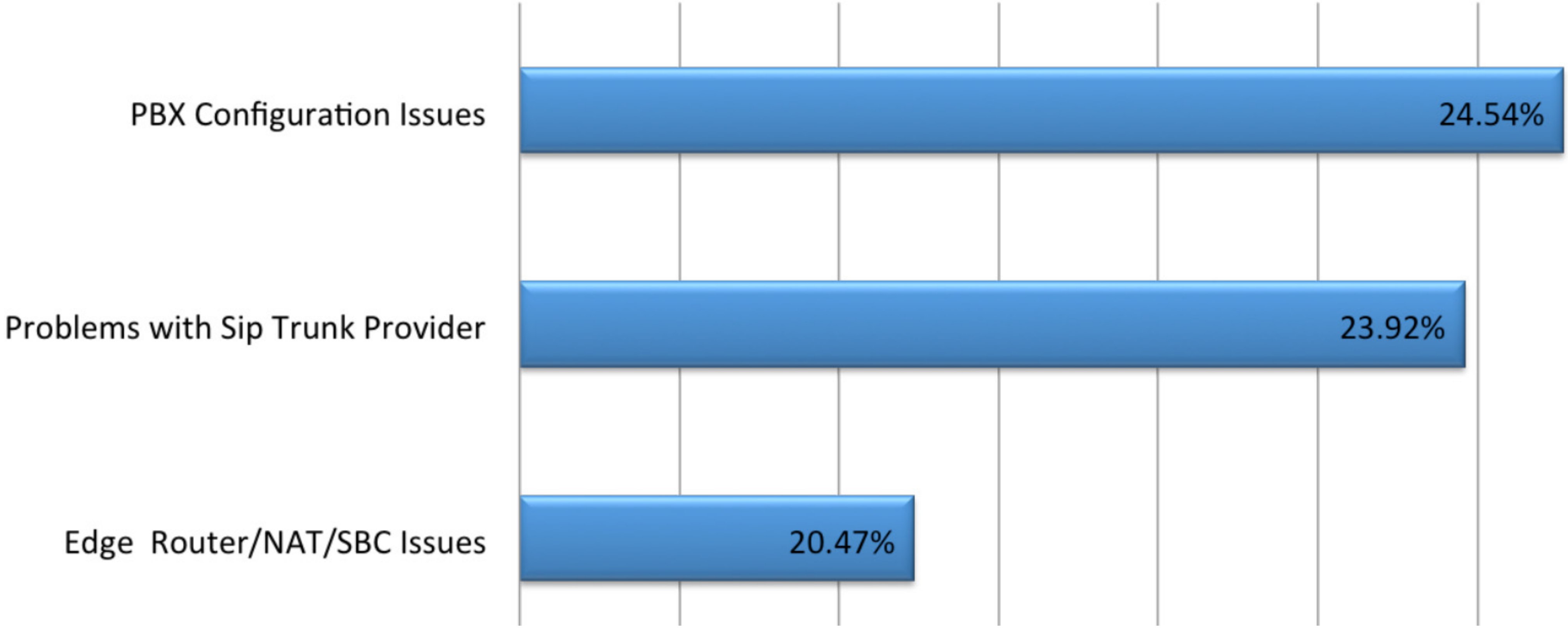
- VoIP.Master **emulates a SIP network** allowing VoIP edge and user equipment to be checked
  - correct operation/performance
  - Up to **30 simultaneous outgoing/incoming calls**
  - Media quality measurement (**E-Model MOS**) for each call and trunk
  - Simple **one button tests** through stored user profiles
  - Comprehensive PDF **test reports** with customisable header/logo
  - Ability to **call different numbers** (destinations), number ranges etc
  - Different **call modes**: Single Call, Sequential Call, Bulk Call
  - Mass Call **mode up to 200 calls** to test VoIP equipment capacity

T.38 Fax emulation



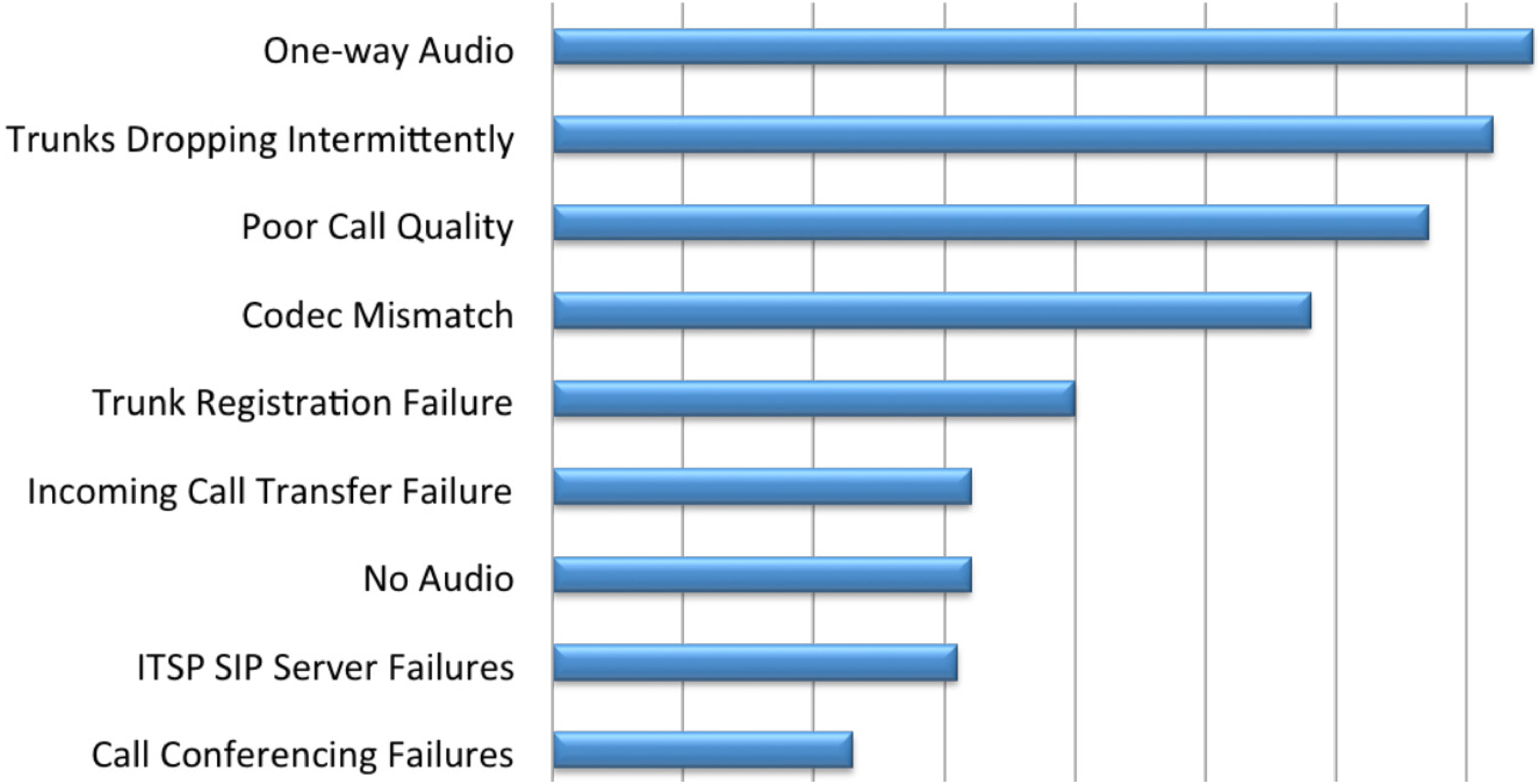
- UC/VoIP.Master can **emulate a T.38 Fax** machine at the customer premise
  - SIP network/trunk to be tested for successful **Fax over IP** (T.38) operation
  - Up to **5 simultaneous outgoing/incoming Fax** calls (requires additional licenses)
  - Support for both T.38 and **G.711 Fax pass-through** modes
  - Received **Fax's automatically converted to PDF** for local viewing
  - Comprehensive PDF **test reports** with customisable header/logo.

Top **Problems** with **SIP** Services



(Source: SIP School – SIP Survey 2014)

Problems with **VoIP/SIP** Trunks



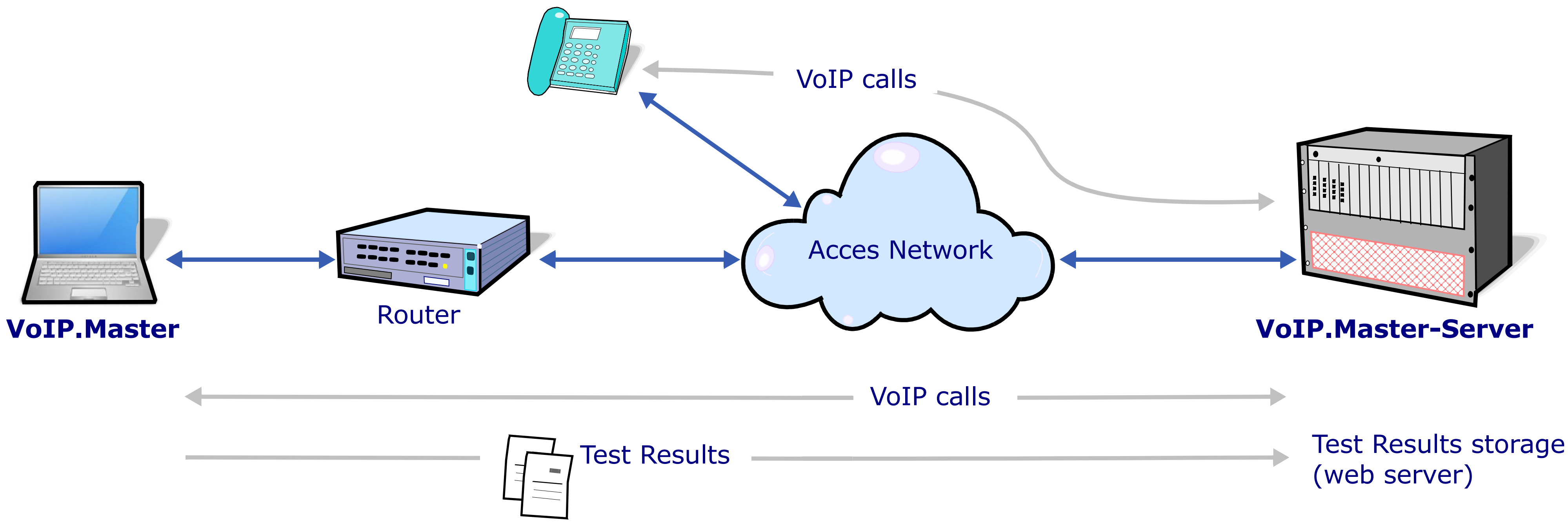
Troubleshooting VoIP

- **Quality** Issues – Check audio paths and call quality (MOS) with different IP QoS settings
- **Codec** Issues – Ensure no codec mismatches between equipment, edge device, and network
- **SIP Registration** Issues – Test that edge device and equipment can register/re-register
- **One Way Audio/No Audio** – Check audio paths (RTP) on all calls (bi-directional) through edge device to connected equipment
- **Calls to PSTN blocked** – Check for ability to call Fixed Line, Mobile, International
- **Router/SBC Lock Up's & Crashes** – Test performance and stability of edge devices under load, number of simultaneous calls and calls per second.

Version Comparison

Feature	*VoIP.Master-lite*	*VoIP.Master*	*VoIP.Master-server*
PBX emulation	yes	yes	n.a.
SIP Network emulation	yes	yes	yes
T.38 Fax emulation	no	optional	n.a.
Terminals in PBX emulation (Simultaneous Calls)	5	30	1000 (server dependent)
Mass Call mode (200 Simultaneous Calls)	no	yes	yes
G.729 Codec	no	yes	yes
Media quality measurement	yes	yes	yes
Calls to PSTN	yes	yes	yes
RTP statistic: (Jitter, Delay, Loss)	yes	yes	yes
DTMF tone	yes	yes	yes
SIP registration	yes	yes	yes
VLAN	yes	yes	yes
TOS/COS	yes	yes	yes
PDF reports	yes	yes	yes
Online software update	optional	optional	optional
Test results storage	local storage	local storage	Central database
Test results access	no	no	yes
ARP and Trace Route	yes	yes	n.a.
Cal Modes	Single, Sequential, Mass	Single, Sequential, Mass	Single, Loop, WAV File (RTP), CLI, Announcement, Call-Back

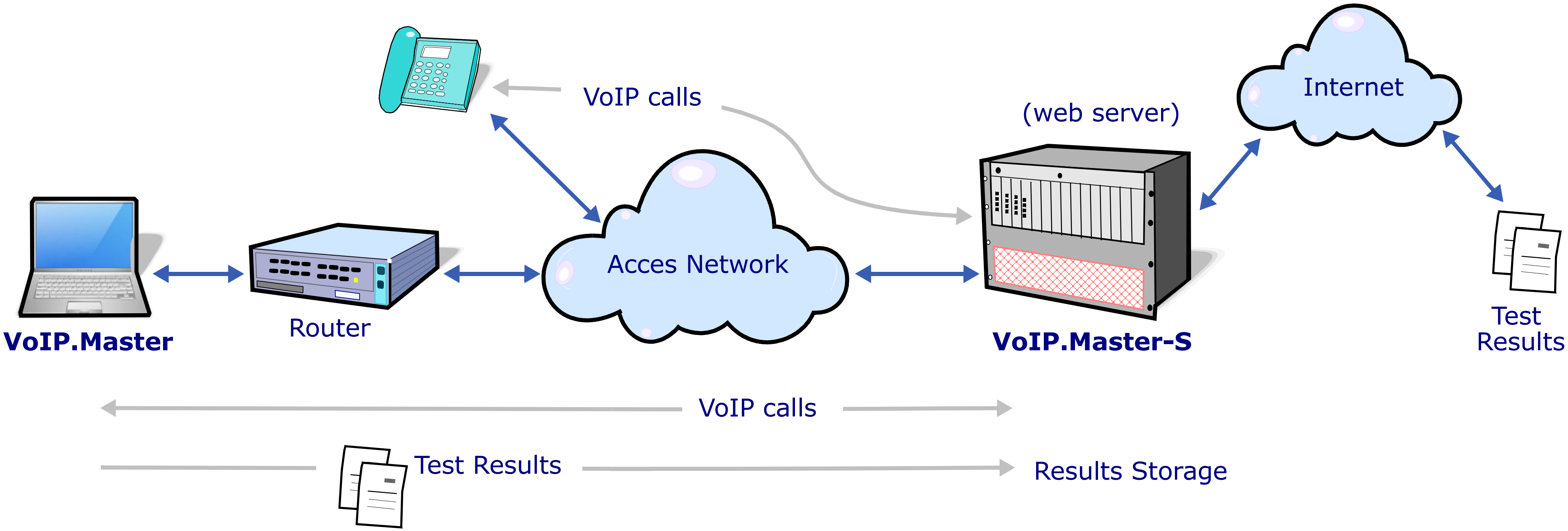
VoIP.Master-Server



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

- VoIP.Master make **test calls** to VoIP.Master-S
- VoIP.Master-S server terminates incoming calls, VoIP.Master signals test mode to VoIP.MasterS
- VoIP.Master-S supports **different modes**: Loop, WAV File (RTP), CLI Announ, Call-Back...
- On completion of test, VoIP.Master **sends test results** to VoIP.Master-S server

VoIP.Master-S: Web Server Interface



- VoIP.Master-S can be used to test Customer Equipment and Network Access.
- Users access VoIP.Master-S through browser using **web server** interface.
- Test results **stored in database** displayed via Web Server; summary and detailed test results.
- VoIP.Master-S operation and configuration controlled through web interface

That's all

ALBEDO Telecom
the Path to Excellence