

Using Open-Source VoIP Technologies to Build a Carrier-Class Telephony Switch

David Zimmer, CEO and Founder

About teresto

Formation	1.1.2002 by integrating of Salink GmbH (part of Xlink/KPNQWest) and Zimmer Medienhaus AG; which had been in existence since 1994.
Headquarter	66663 Merzig, Trierer Strasse 223-225, Germany (approximately 30km from City of Luxembourg)
Shareholders	VSENET GmbH, Saarbrücken; 75,5% (RWE Group) Zimmer & Associates GmbH, Merzig; 24,5%
Management	David Zimmer, CEO Tobias Zimmer, COO Monika Gross, CFO
Revenues	2005: 4,9 Mio EUR 2006: > 6 Mio EUR (planned)
Employess	23 total

Our little Voice (-over-IP) history

Date	Event/Milestone
2000	Our parent company -VSENET- invests in multiple TDM-only Siemens EWSD switches, largest single investment ever
2002	teresto buys Cisco CallManager PBX; project becomes one of our worst nightmares in technical as well as commercial terms within teresto's corporate history.
2003 - 2004	Testing and playing around with open-source VoIP tools like GNU Bayonne, SER, Vocal, sipX and early version of Asterisk
October 2004	teresto is assigned VoIP project lead within VSENET since VoIP „is an IP issue“ according to the TDM voice folks
March 2005	Start of the VoIP SoftSwitch development project within teresto.
Late 2006	Official start for our VoIP product offerings with SLA and such.

VoIP: more than other transport ...

- It's Mainframe vs. Client-/Server
- Closed, proprietary solutions vs. open standards
- Central authority vs. distributed networks, maybe sometime even peer-to-peer
- Limited choice vs. total freedom
- Voice becomes just ONE service like any other on a given IP network

teresto VoIP platform in general

- Layered approach
- „dumb“ nodes
 - ◆ intelligence resides within distributed dialplan from real-time database
- Commodity hardware (Zapata approach)
- Currently 100% Asterisk based, but open for other products as well
- Real-Time Architecture with MySQL Cluster
- DUNDI

Layered approach

Existing PSTN Network
other Carriers, i.e. VSENET, Deutsche Telekom, Telefonica

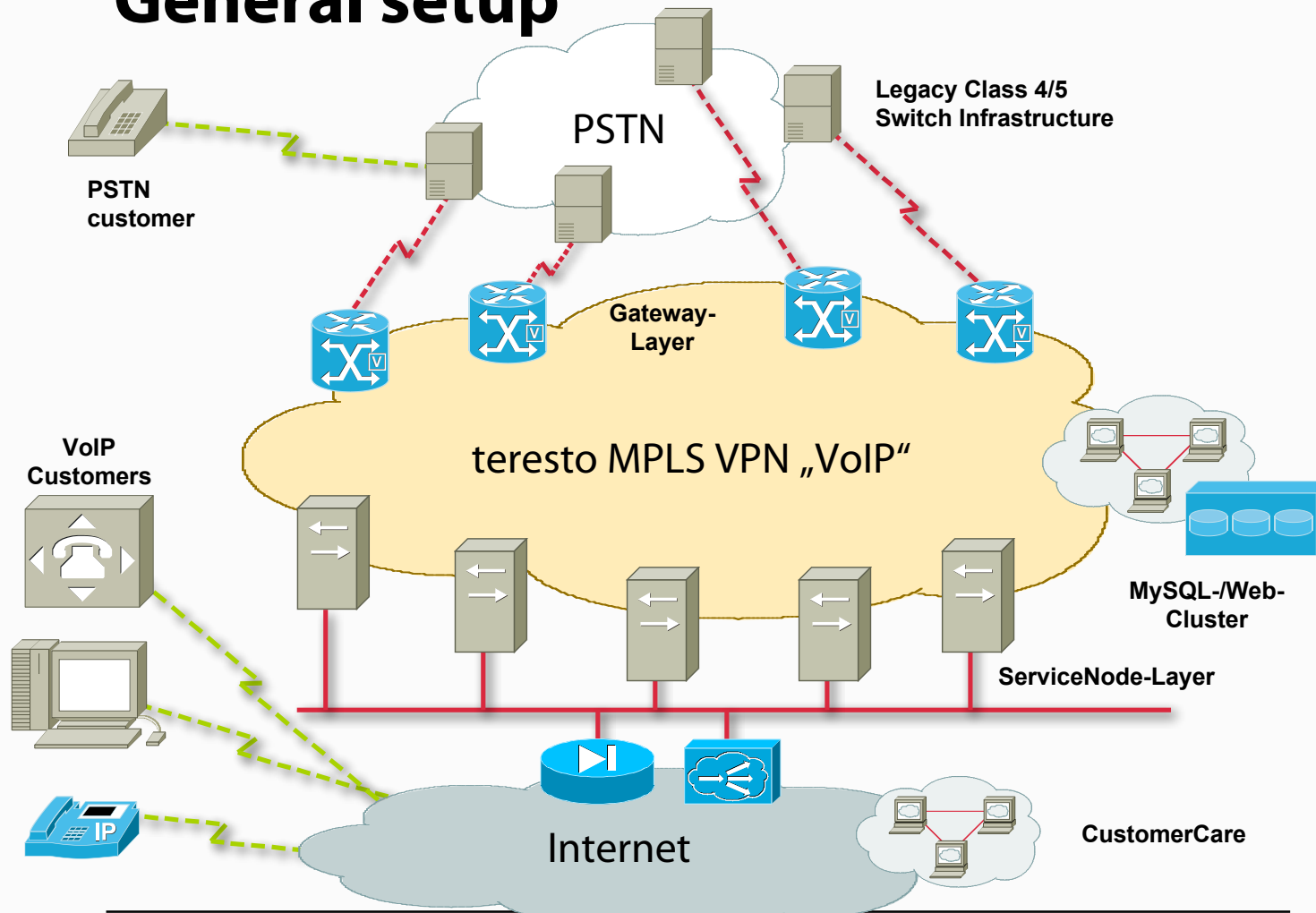
Gateway-Layer, w. and w/o. Conversion
(SS7, ICAs, G.711, Least-Cost-Routing, SIP Trunking)

ServiceNode-Layer (Registrar, Proxy, SIP, IAX, H.323, MGCP,
ENUM, VoiceMail, Fax-Services, Conferencing, CDR, IVR)

Process- und Billing-Layer (MySQL, Least-Cost-
Logic, CDR Reporting, Supplier-API, Purchasing Control)

CustomerCare- and Provisioning-Layer (Web-Portal with
different views for customers and employees)

General setup



Gateway layer

- SS7 Signaling to PSTN with solution from Cosini Technologies (commercial)
- E1 trunks to PSTN with Digium and Sangoma hardware (with echo cancellation)
- SIP trunk to PSTN
- IAX trunk to PSTN
- GatewayNodes
 - ◆ Debian Linux
 - ◆ Dual Xeon @ 3 Ghz
 - ◆ 2-4 GB RAM
 - ◆ 120 - 240 concurrent sessions

ServiceNode layer

- Each Service runs on separate box no „all in one“ approach
- Services
 - ◆ SIP Registrar/Proxy
 - ◆ IAX Registrar
 - ◆ MGCP/H.323
 - ◆ Instant Messaging
 - ◆ IVR
 - ◆ VoiceMail
 - ◆ Any ideas
- Dialplan not stored locally but with real-time architecture in MySQL cluster
- Extension discovery is done through DUNDI
- If DUNDI yields no result, calls will be tranfered to Gateway-Layer for further processing
- Commodity hardware
 - ◆ Pentium 4
 - ◆ 1-2 GB RAM
 - ◆ FreeBSD 5/6

Load-Balancing

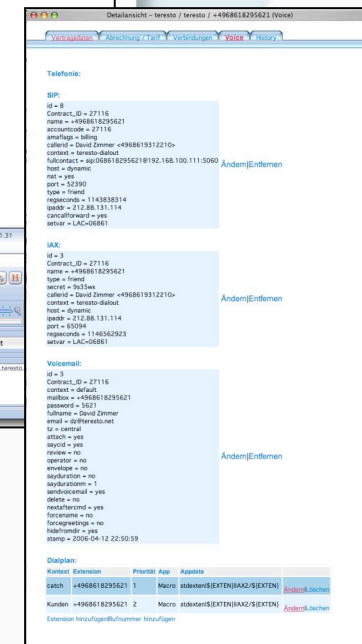
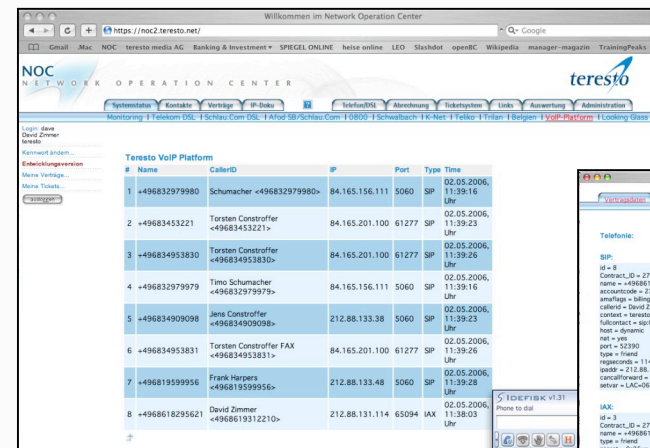
- Via Linux VirtualServer (LVS) project
 - ◆ <http://www.linuxvirtualserver.org/>
 - ◆ We use the „DirectRouting“ mechanism
 - ◆ Seems to be least NAT problematic
 - ◆ Currently not usable with Asterisk's SIP stack :-(but should work with next major release (1.4)
- Other products tested as well
 - ◆ Vovida Load-Balancer
 - Layer7 load-balancer
 - Asterisk does not handle the keep-alive packets
 - Does not integrate well in NAT environments
 - ◆ Cisco Content Switch (CCS) is being tested now

Process- and Billing-Layer

- All configuration is kept in a MySQL database
 - ◆ sip.conf
 - ◆ iax.conf
 - ◆ extensions.conf
- For highest availability we use a MySQL-Cluster
- Asterisk accesses the required tables through ist „Real-time architecture“
- Standard interfaces SOAP, ODBC etc. to partners for LNP, Billing, Sourcing etc.
- Outbound Routing intelligence
 - ◆ Quality issues
 - ◆ Source routing
 - ◆ Least-Cost-Routing

CustomerCare and Provisioning

- Web-Interface for customers and employees to provision services
- Real-Time interface to the „Softswitch“
 - ◆ Asterisk Management Interface (AMI)
 - ◆ AstManProxy to query multiple * servers
- Real-Time CDR Interface



Areas for improvement

- SS7 extension for TCAP and other protocols
- Faxing
 - ◆ T.38 client and server implementation needed for ServiceNode-Fax
- Asterisk internal
 - ◆ Multi threaded IAX stack
 - ◆ SIP binding to specific ports and IP addresses for
 - ◆ Support for Wideband-Codecs
- Dundi
 - ◆ more granular metrics for selection of destination (i.e. borrow some ideas from BGP route selection algorithm)
 - ◆ Make DUNDI an RFC :-)

Open issues at teresto

- Much more load testing
- Fully integrate ENUM into platform
- More intelligent gateway selection based on real-time performance metrics on gateways
- AGI scripting, AGI scripting, AGI scripting
- ServiceNode virtualization with VMWare or Xen
- Exploit TDMoverEthernet-Feature (TDMoE) of Asterisk to build a „ServiceNode-TDM“ and thus a pseudo-TDM switch

Wrap up

- Open-source tools can be used to build carrier-class voice solutions
- Paradigm shift necessary
 - ◆ Not one big box that does it all
 - ◆ Rather a swarm of dumb boxes networked together
 - ◆ No single supplier that can be blamed for failures and errors
- Asterisk has grown from humble beginnings to a major force in private as well as public switch environments.

Questions

Thank you.

