Building the NGN Switch (with Asterisk)

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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%
Business	STM-16/OC-48 based MPLS-/IP-Backbone in the heart of Europe, connecting Frankfurt, Luxembourg, Brussels and other major central European cities.
Revenues	2005: 4,9 Mio EUR 2006: > 6 Mio EUR (planned)
Employess	23 total, ISO9000:2001 certified

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 the 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.

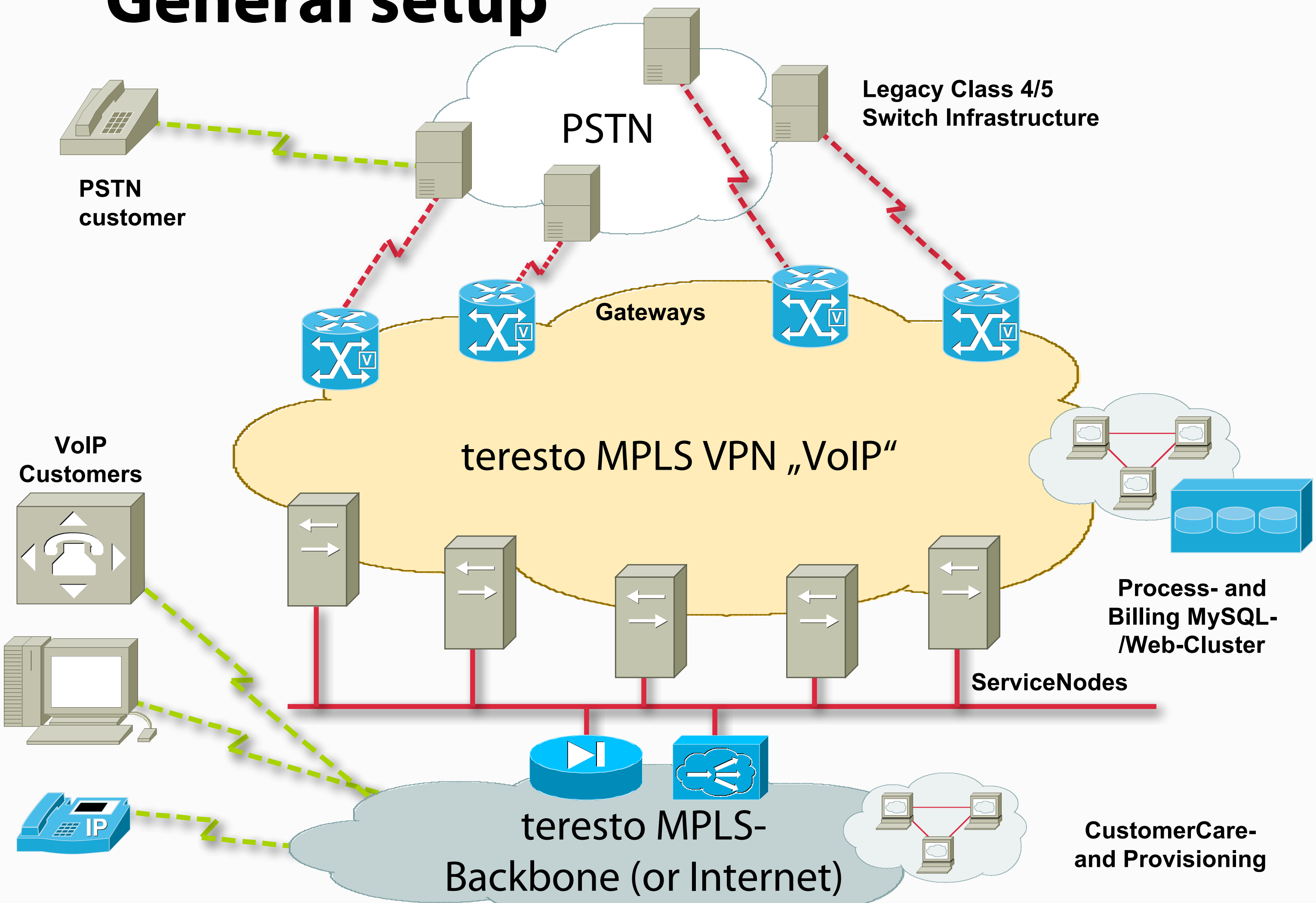
teresto VoIP platform in general

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

Layered approach

Existing PSTN Network other Carriers, i.e. Verizon, Deutsche Telekom, Telefonica
Gateway-Layer, w. and w/o. Conversion (SS7, ICAs, G.711, Least-Cost-Routing, SIP Trunking, SBC)
ServiceNode-Layer (Registrar, Proxy, SIP, IAX, H.323, MGCP, VoiceMail, Fax, Conferencing, 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 -> change to P4 or comparable
  - 2-4 GB RAM
  - 120 - 240 concurrent sessions

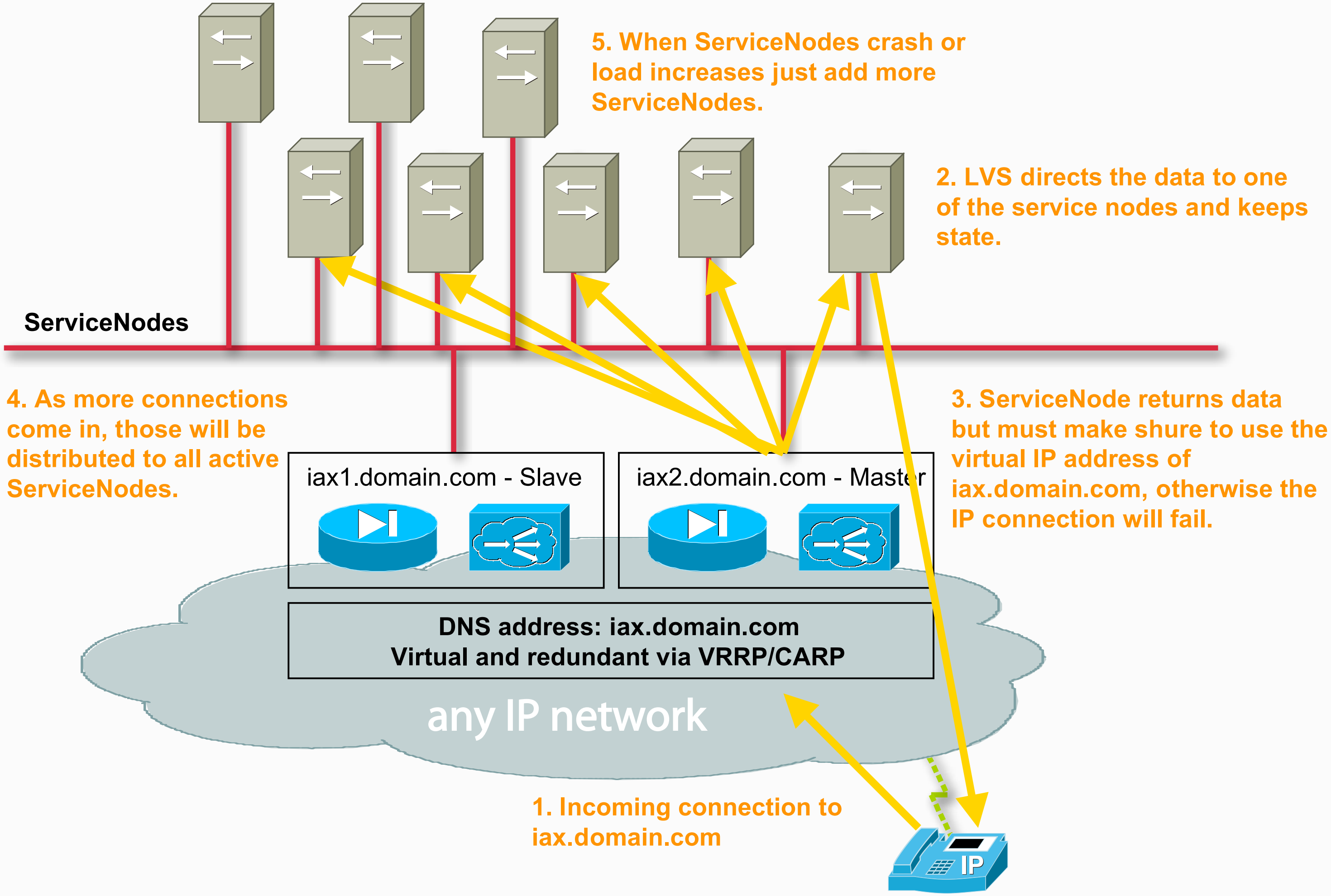
ServiceNode layer

- 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 other 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
- Hardware
  - P4 -> DualXeon
  - 1-2 -> 4 GB RAM
  - FreeBSD 5/6 -> Debian
- New: Transcoding on ServiceNode

Load-Balancing

- Via Linux VirtualServer (LVS) project
  - http://www.linuxvirtualserver.org/
  - We use the „DirectRouting“ mechanism
  - 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)

Load Balancing with LVS/DR



Process- and Billing-Layer

- All configuration is kept in a MySQL database
  - sip.conf
  - iax.conf
  - extensions.conf
- For highest availability we use 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
  - Static call Routing
  - Quality based Routing
  - Least-Cost-Routing

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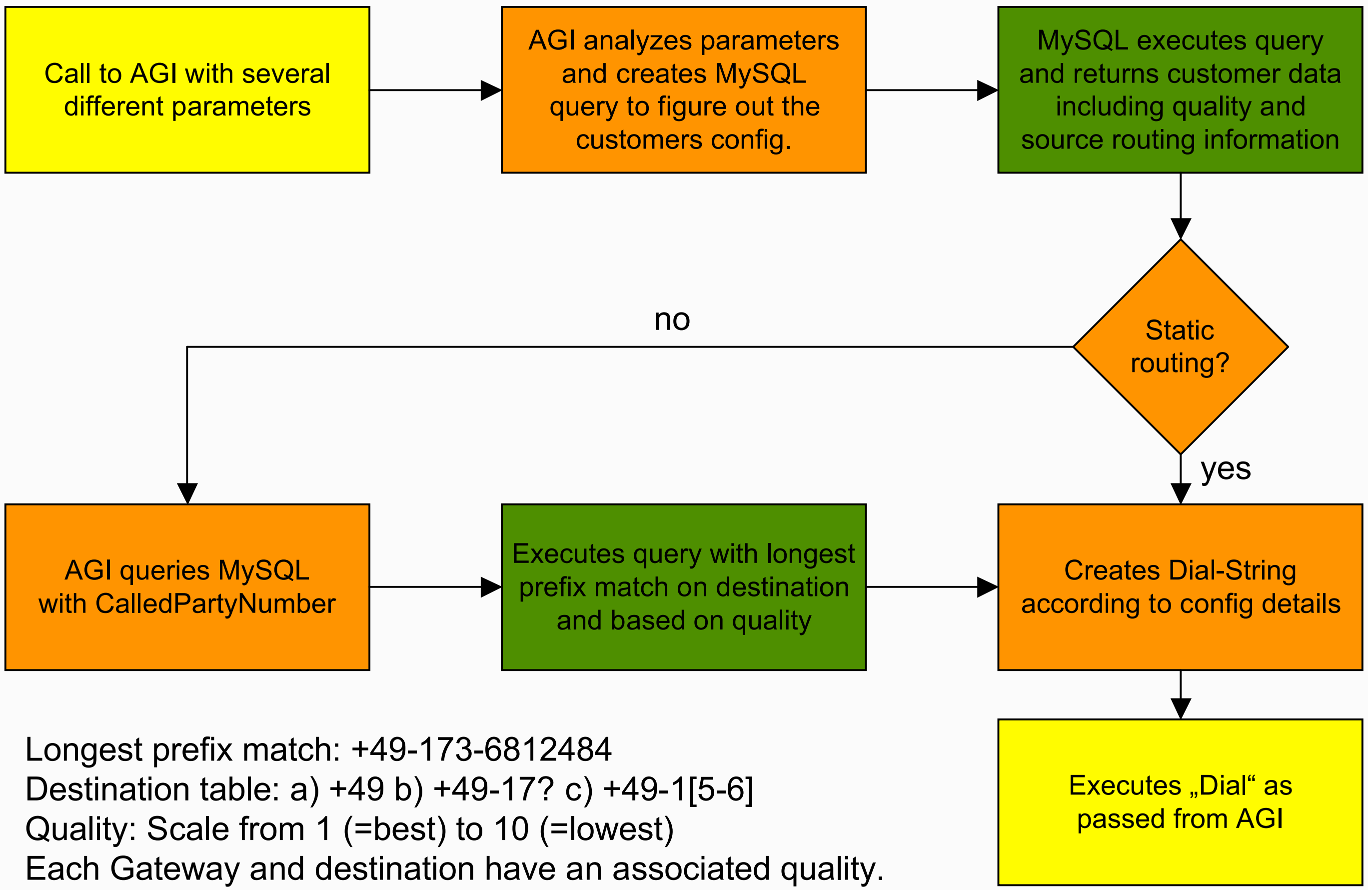
Outbound Routing Intelligence



Longest prefix match: +49-173-6812484
Destination table: a) +49 b) +49-17? c) +49-1[5-6]
Quality: Scale from 1 (=best) to 10 (=lowest)
Each Gateway and destination have an associated quality.

Dialplan

AGI

MySQL

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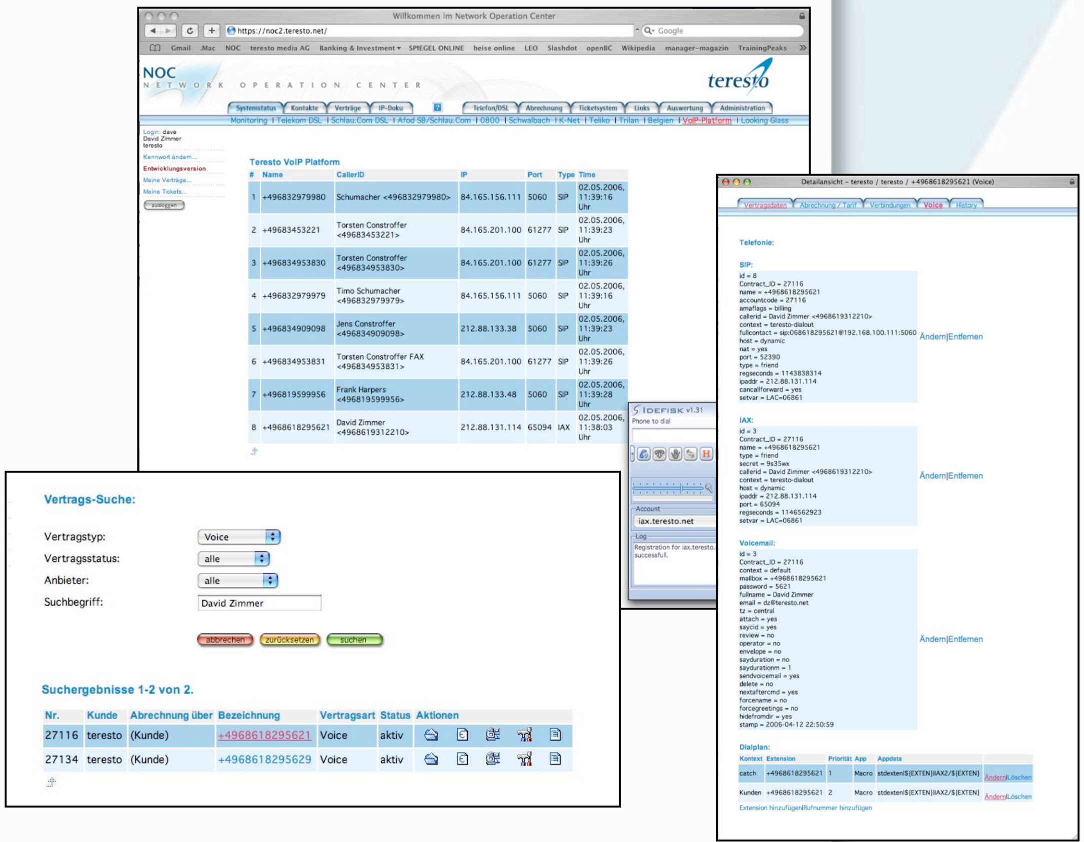
CustomerCare and Provisioning

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



Areas for improvement

- SS7 extension for TCAP and other protocols (SMS)
- Faxing
  - Stable T.38 client and server implementation needed for ServiceNode-Fax
- Asterisk internal
  - Multi threaded IAX stack (1.4)
  - SIP binding to specific ports and IP addresses for (1.4)
  - Support for Wideband-Codecs (1.???)
- 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 enhancements
- ServiceNode virtualization with VMWare or Xen
- Exploit TDMoverEthernet-Feature (TDMoE) of Asterisk to build a „ServiceNode-TDM" and thus a pseudo-TDM switch
- Work towards future legal requirements

Legal Requirements (CALEA, TKG)

- Emergency Calling
  - Easily solved within the dialplan
- Call interception and monitoring
  - We use monitor() application in the dialplan
  - Separate Web-/FTP-Server for law enforcement officials to trigger the monitoring and download the rtp data
  - Requires the media (rtp) path through our network
- Call tracing
  - SS7 protocol gives us all the required call information to do this

Wrap up

- Asterisk and other open-source tools can be used to build carrier-class voice solutions
- Paradigm shift necessary
  - Not one big box that does it all
  - Rather many dumb boxes networked together
  - No single supplier that can be blamed for failures
- Carrier switching works very stable since very few features are actually used!
- Our concepts can be applied to most large scale Asterisk deployments

Questions

Thank you.

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