Challenges and Lessons Learned in Implementing SIP–based Communication Services

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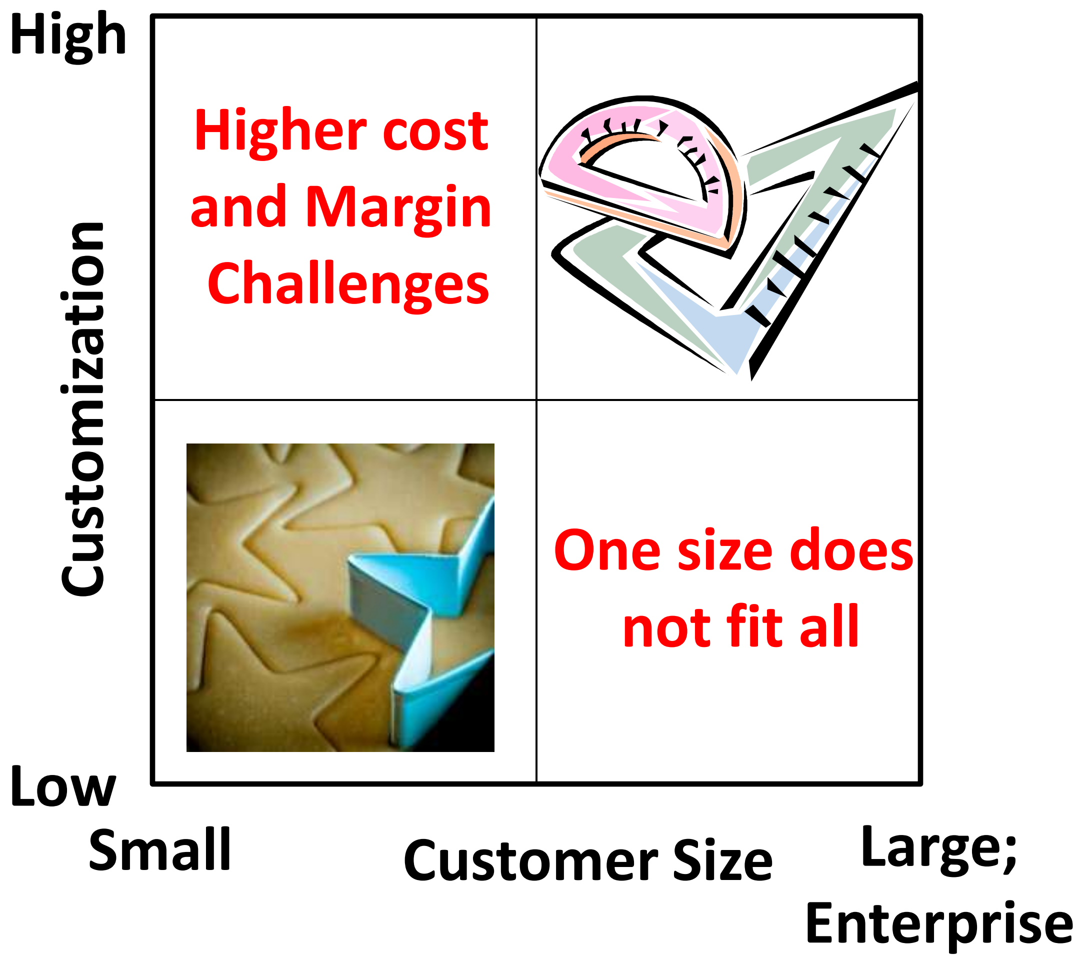
Outline

- Based on what we went through and learned
- May not fit every SP but worth considering
- Not just technical issues but holistic approach
  - Interoperability
  - Support
  - Capacity Planning
  - Standards upgrades
  - Technical
- Best Practices for Upcoming new capabilities ?

Interoperability Lessons

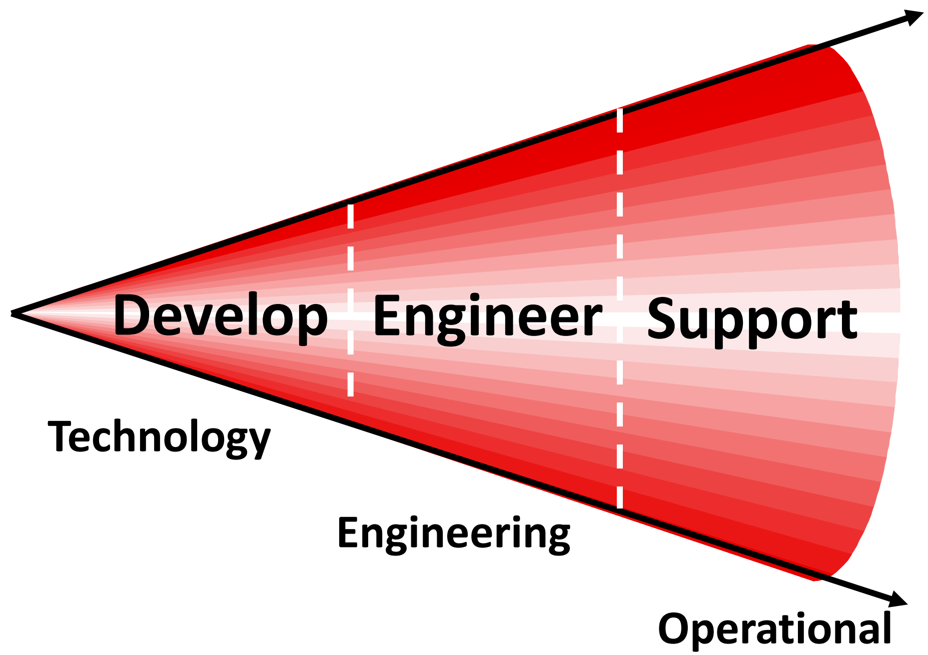
- Phasing PBXs for validation
- Methodology
- Lab validation or production network trials
- PBX / premise SBC combinations
- Open source PBXs
- Scaling and sustainability – “Mass Customization” ?

"Mass Customization" - Oxymoron or Attainable Service Provider *Nirvana* ?



Support

- Internal support structure
- Product Lifecycle LOE
- Training
  - Modes
  - Knowledge domains
  - Sectionalization
  - Tools
- Vendor support structure



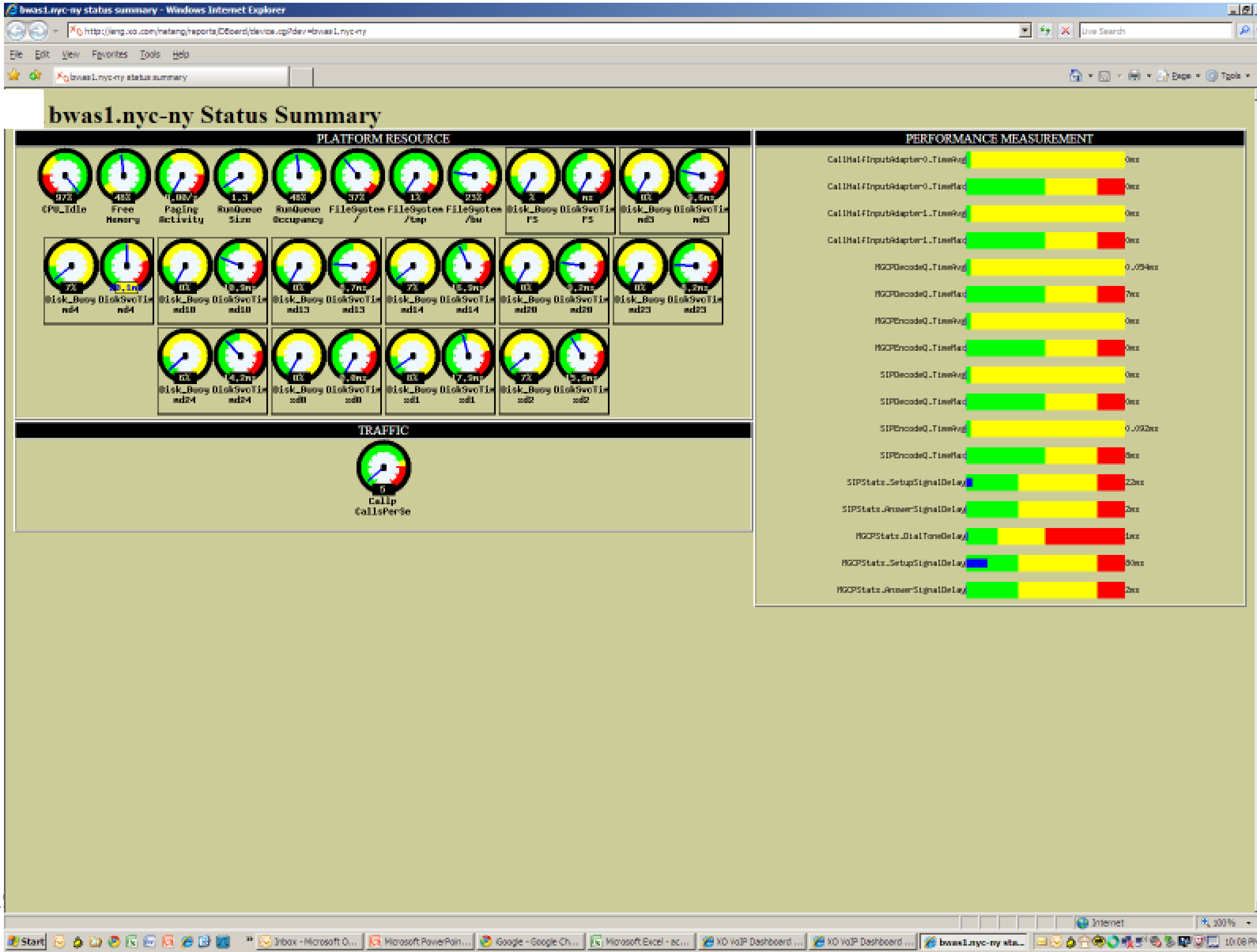
Capacity Management Lessons

- Idle capacity vs. opportunity cost
- Demand forecast challenges
- Some subsystems may have long lead time
- Priming network with initial capacity
- NEs dedicated to new product in the early stages
- Adapt to actual demand growth and share NEs for many products
- Overcapacitize, segment and monitor platforms
- Overload controls and congestion management

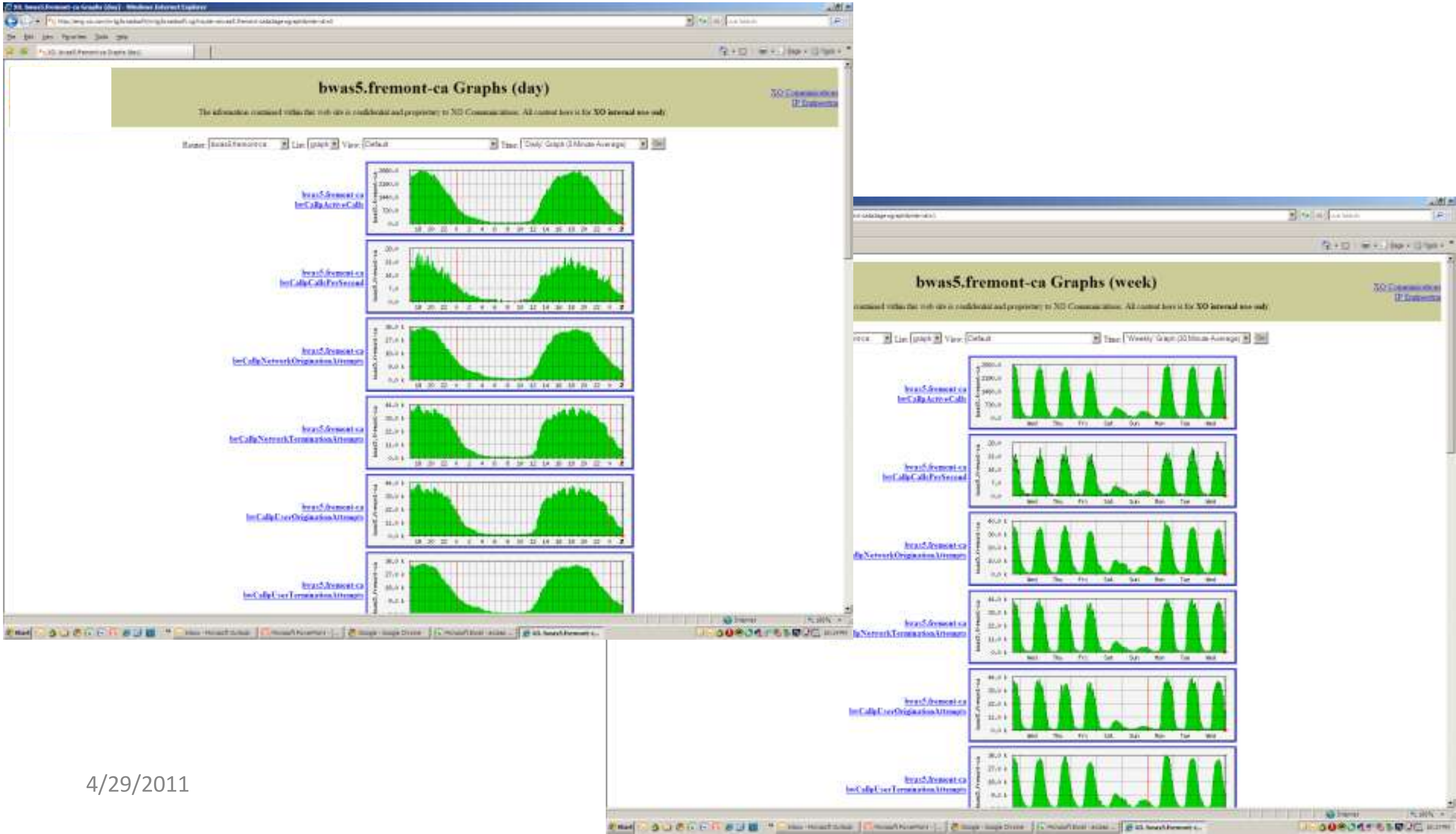
Standards and Implementations in the Context of Commercial Reality

- Dependency on vendor’s velocity
- Large networks with high automation; Best-of-breed multi-vendor networks
  - Significant effort and time for regression tests
- Risk of premature implementation of work-in-progress standards practices
- Cost vs. benefit trade-off ; batching changes
- Backward compatibility in standards and allow time for SPs to roll out

Dashboard


bwas1.nyc-ny status summary - Windows Internet Explorer
bwas1.nyc-ny Status Summary
PLATFORM RESOURCE
PERFORMANCE MEASUREMENT
TRAFFIC

MRTG PM Charts



Technical - Lessons

- Redirections
- Mid-call negotiations
- Codecs
- T.38
- Unreachability

Redirections

- Some PBXs Forward calls with PSTN numbers in the 2nd leg
- NE verifies users; call blocked; Diversion Header
- Initially some PBX vendors did not implement Diversion Header but only History Information Header
- Not all NE vendors had implemented History Info header
- Now these vendors also support Diversion Header
- Many customers ask for REFER
- Different PBXs implement differently; interop issues ; so we test in detail
- Ensure mediation and billing of redirection CDRs
- Special attention to Redirections during certification and trials

Mid-Call Negotiations

- Inbound call presented via premise to Customer’s IVR who only could do G711
- The call was transferred to an endpoint which requires G729a
- Premise did not do a clean transfer
- Endpoint saw some training G711 packets and the G729a packets
- Premise saw the change in RTP sequence numbers and interpreted it as packet loss and one-way audio was resulting
- Although interface may be ok started testing all the high runner use cases end-to-end

Codecs

- When retail service started most traffic was from / to PSTN
- Backbone should be of the highest quality and simple – so only G711, no SS, DTMF in audio ; bandwidth was not a big consideration
- Made sense to transcode access codecs to normalized G711 into the backbone at the edge of the network in the SBCs
- Newer services like HD Voice may need a new strategy ; since it is not yet ubiquitous offer may be only on-net. But we want the calls to gracefully fall back to G711 instead of failing; and no *a priori* knowledge from the TN HD capability we want end-to-end negotiations as opposed to transcoding at the edge.
- Also SBCs may not have transcoding capabilities for all the new codecs
- Deliberately chose to have transcoding in the beginning and introduce negotiations now
- SPs have to conscious decisions synced up with their service roadmaps

T.38

- Fax only customers want to start with only T38
- Different implementations
- Criteria for triggering T.38 vary between implementations
- Some times deadlock on who has to initiate T38, although in theory Receiving Gateway should
- Negotiations take time , especially G3 – SG3 if started as G729a call and fax machines may time out
- Multi-page transmissions sometime fail, in spite of clean IP and slip-free TDM
- Very cautious about T38 and test every customer opportunity
- Noted that SIP Forum FOIP is grappling with the issues and working with ITU to get resolution

Unreachability

- Different vendors (PBX , NE) implement unreachability detection and Service Resilience differently
- Some use failure to receive proper response for an Invite which checks service availability
- Some use frequent SIP OPTIONS could indicate SIP stack is OK but may not convey Service Availability
- Stateful (this call and new calls are rerouted) vs. stateless (every call will go through the same protocol Timeouts) ; if stateless for high probability events PDD is increased
- Concern about erosion of cps capacity due to the number of messages to be handled for a call (Session Refresh Timer , complex call flows etc.)
- SPs may cap frequency of OPTIONS from premise

Upcoming Areas for Developing Best Practices - Enhanced Capabilities ?

- New media and Multi-media capabilities
  - HD voice; going beyond on-net
  - Interactive video communications
- Unified Communications
  - As “adjunct” in the premise
  - Shared and hosted in the cloud
- Mobility
  - FMC (macro, wireline, premise wireless)
  - Unified messaging

Conclusions

- Efforts of SIP Forum valuable for industry consensus
- Kudos to all those dedicated folks engaged in the effort
- Industry has made a lot of progress
- Array of new technologies and services coming up
- Reward for good work is more work !!

Thank you

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