TechWiseTV Workshop
SIP Trunking
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SIP Trunks for PSTN Access

Perception vs. Reality

Perception	Reality
SIP Trunks can be deployed over any media	▪ SIP Trunks should only be deployed over media that can provided a guaranteed QoS that is acceptable (i.e. it would not be recommended to deploy them across Satellite links if Voice quality is important)
SIP Trunks are always cheaper than PSTN trunks for PSTN Access	▪ Large Enterprise have such low rates for traditional TDM based telephony, rates over SIP Trunks may not save much, if anything, in per minute charges for Local or Long Distance voice calls ▪ SIP trunk are unregulated services and what SP charge vary widely (unlike TDM offerings) ▪ If cost benefits do accrue for a customer, it is likely operational or equipment costs, not service costs
SIP Trunks provide the exact same experience for the end users	▪ SIP Trunks can provide the same experience in many cases, but some cases (i.e. Baudot connections for Deaf users or V.92 speed modem connections) experience is different; fax can be problematic
SIP Trunks are easy to deploy and just work	▪ SIP is easy to deploy, but interconnection between different vendors implementations of SIP and different Service Providers offering is not yet ironed out ▪ Current SP offerings are not mature and every provider's offering has to be carefully evaluated and tested ▪ Number portability can be a significant enabler or drawback
SIP Trunks should always be used	▪ Evaluate carefully. In some cases TDM trunks make a better choice. Or perhaps a better choice for certain traffic types all call patterns.

Interoperability Issues

- There is currently no standard for SIP Trunks that can provide the same level of consistency and interoperability of PSTN ISDN Trunks
- There are efforts underway in the industry to have more interoperability; various efforts are being lead by the SIP forum, ATIS, TISPAN
- The problem of interoperability is reduced by having a customer owned border element that can provide signaling interworking and transcoding
- This problem can be further reduced by having a Service Provider owned Border Element that acts as a demarcation point for signaling
- Customer should test, test, test before deployment of their first SIP Trunks solution, and replicate after that for scaling
- Use SIP Profiles to TWEAK signalling
- Do not RE INVENT the wheel, replicate what is easy

Quality of Service (QoS)

- Quality of Service is essential for ensuring that a TDM trunk can be replaced with a SIP Trunk for PSTN Access
- QoS can only be GUARANTEE when Physical link (ie Layer 1 Provider) is the same as SIP Trunk (ie Layer 7 application provider).
- Good QoS policy can help (ie marking packets), when Layer 1 provider is same as SIP Trunk provider (ie put your voice traffic in correct Queue) but if you choose an “over the top” SIP Trunk provider, marking of QoS packets will have little effect on the final quality provided.
- Takeaways;

  Mark packets, but to not trust that marking packets will ensure quality

  When using “over the top” providers, realize that marking will not be honored.

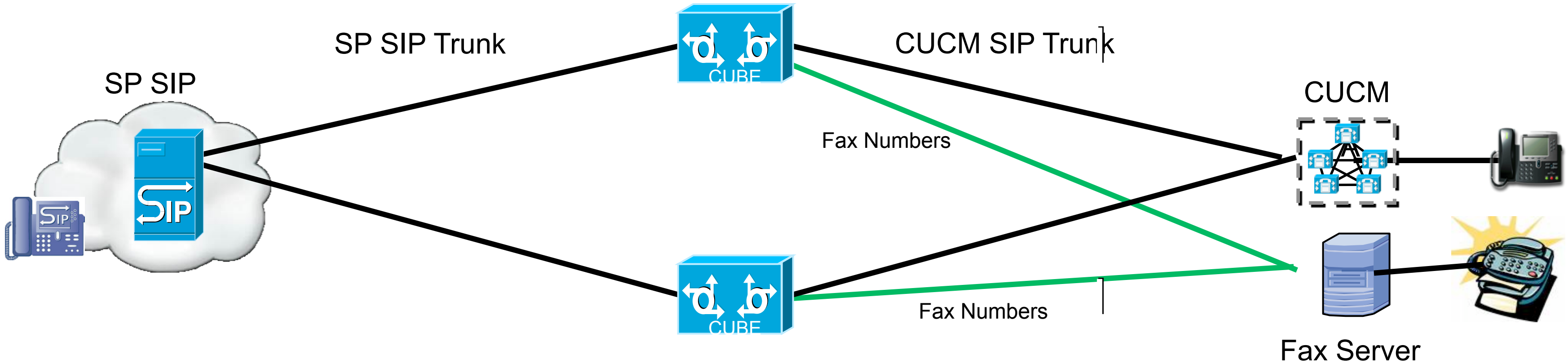
Fax Calls

- SIP Trunks can typically use three different methods to supports FAX calls
  - All calls are sent as G711
  - Call sends a RE-INVITE to up-speed to G711 when a FAX tone is detected
  - T.38 FAX capabilities are exchanged and fax relay is used
- SIP Service provides also occasionally offer a separate fax to -mail service using T. 37 Store and Forward fax
- T.38 to Inband FAX (i.e., Fax over G711) will not result in acceptable CSR

Fax Method	T.38 Fax Capabilities Exchanged as Part of SIP Messages	All Call Sent as G711	Fax Tone Is Detected and RE-INVITE to up-speed to G711 Is sent
Pros	▪ Highest fax success rates can be achieved ▪ Cleanest solution from signaling and media point of view ▪ Use less bandwidth than G711 ▪ Fax and Voice calls differentiated	▪ Most widely deployed ▪ Simplest solution	▪ Provides benefits of least bandwidth with G729 call initially upspeeding to G711 if call is FAX ▪ Tone (2100Hz) can be mixed between Modem and Fax ▪ Fax Pass-Through
Cons	▪ Degree of interoperability ▪ Not offered by many Service Providers	▪ Consumes a large amount of bandwidth for all calls ▪ No ability to distinguish FAX calls from Voice calls in CDRs	▪ Each vendors support of RE-INVITEs is different ▪ Currently not supported with all Cisco equipment

Fax Servers and SIP Trunks

- Fax Servers generally are deployed at head end locations to provide a centralized fax offnet/onnet for many locations
- Fax Servers will have a LOWER SUCCESS RATE when using a SIP trunk then a PSTN trunk at the fax server
- Fax Servers interoperability with SIP Trunk services are getting better, HOWEVER, most Fax Servers work best with T.38 and many SIP Trunk services do not offer T.38
- Options
  - Leave TDM trunk in place for Fax Servers
  - Accept lower fax success rate with PROCEDURAL workaround for failed faxes
  - Route all calls to CUBE and then have CUBE look at called number and route Fax calls to fax server and other calls to CUCM



Supplementary Services

- The supplementary service invoked over the SIP Trunk is not supported or understood by the far end SIP switch

  For example, the signaling to place a call on **hold** and temporarily stop media can be done in one of several ways, all of them are compliant with the standard; mismatching methods may be supported between two SIP switches

- Testing of Supplementary Services before deployment is only way to ensure success

  Create a test case for each service before deployment

  Report findings to Service Provider

  Determine if lack of these functionality should effect deployment

- Typical Supplementary Services test cases

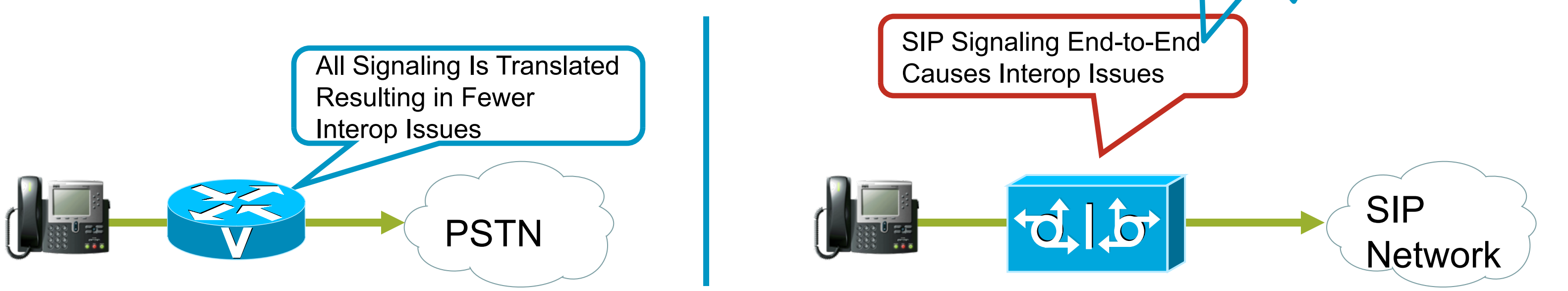
  Placing call on HOLD

  Forward on Busy/No Answer to Number within premise

  Transferring call to another extension

  Correct billing for forwarded calls

CUBE1.3 resolves many interop issues



Fraud Issues

- Only accept INVITE from people you know
    - Use ACLs
- Only send INVITE to people you know
- Review your SIP Trunk bills
- Disable international calls by default (do not use ".T" dialpeer)