BRIEF FROM INTERTEX DATA AB AND INGATE SYSTEMS AB

By Karl Erik Ståhl, March 23th, 2012

Read more at http://internetplus.intertex.se Also see presentation:
http://www.ingate.com/files/itexpo_miami_2012/Intertex-Overview_of_an_Internet+_Model.pps

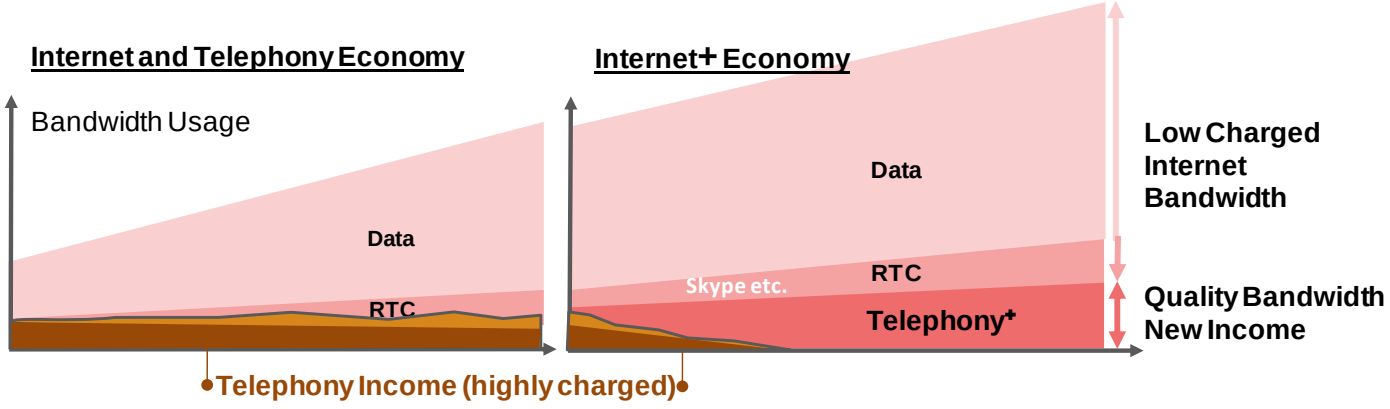
Internet+

The Way Toward Global Unified Communication and Replacing the Telephone Network

Extend the Internet a bit, apply Internet thinking, provide a standard real-time interface for users and servers and we can get:

The bandwidth and network elements already exist and the additions needed only cost a fraction of what is spent on POTS replication over IP today.

In the proposed Internet+ model, network service providers are again given a way to earn on good offerings to users, getting revenue by providing quality bandwidth (instead of stopping VoIP over wireless and similar).



Internet+

Internet+ is high quality real-time communication brought to the Internet, with metering and trust to allow new services as well as including the plain old telephony service (POTS) and replacing the PSTN.

It is time to release the power of our global network, our terminals, PCs, smartphones and having old telephony (POTS) as a seamless part thereof. The components are available, can replace the PSTN, can extend the Internet and will make the costly POTSoIP structure with all its problems obsolete.

In addition to getting a global real-time multimedia-competent network, at higher reliability and vastly lower infrastructure cost than IMS or other types of SDN networks, service providers can both replicate old business models as well as apply new business models.

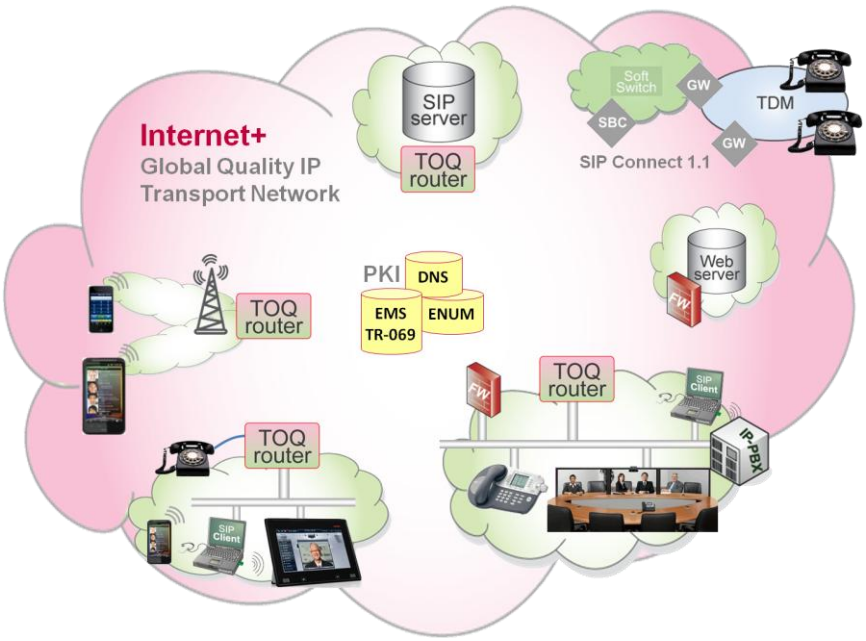
The keys to an Internet+ world of communication:

A non application-specific transport network: Just like the Internet!

- \+ Prioritization for real-time traffic - Just enable diffserv!
- \+ Delivery to the users, on LANs and to smartphones
- \+ Metering and charging of “beyond Internet usage”

For a Telephony+ service (including POTS):

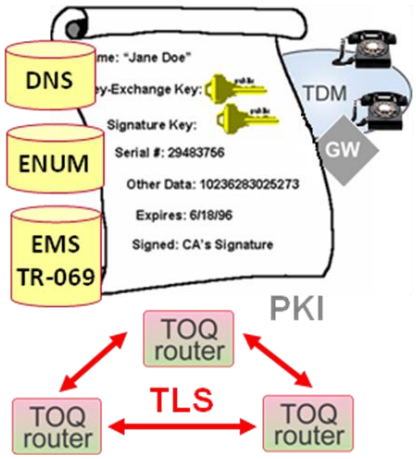
- \+ SIP being the standard to use (no deviations, extensions or gateways inside the global network!)
- \+ Usage of E.164 numbers in addition to SIP address (ENUM to convert to SIP or even to Skype and other islands)
- \+ Trust between participants (like having a telephony subscription/telephone line/number today)



The Internet+ can be viewed as an Internet access where priority/QoS is also available. The TOQrouter includes functions found in competent E-SBCs, such as a SIP proxy that both solves the firewall/NAT traversal problem and routes calls or messages directly to their destination, without introducing incompatibility and degeneration. It also classifies traffic for prioritizing - QoS. The TOQrouter preferably also contains the SIP registrar, metering and CDR generation for real-time telephony-type services. The interface is SIP as defined, allowing all SIP-based services, without introducing deviations.

Calls to and from PSTN-connected phones are routed to the service provider’s gateway closest to the PSTN-connected party, which may be an SBC/softswitch-overlay network with a SIP Interface.

SP Infrastructure without softswitches and SBCs



The Internet+ model, allows trust, billing, and management to be separated from the IP transport of a certain service, thus benefitting from the advantages that made the Internet so successful.

For providing the old telephony service as well as any real-time multimedia service (Telepresence, IM, presence, SMS etc.), the service provider’s infrastructure collapses to commonly used elements such as DNS, ENUM, and Management System for the “TOQrouter”.

TOQ stands for Trust, Open, Quality, and the TOQrouter includes a SIP proxy used for routing calls and messages directly between the endpoints as well as for delivering to users on the LANs or to 3G/4G phones (resolving the firewall/NAT traversal issues without use of work-around methods).

Service provider issues CA-signed certificates for trust, using mutual TLS between all SIP proxies on the WAN side. The same certificates can be used for the meter and to deliver CDRs for billing. The proposed TR-069 management is highly scalable, secure and excellent for reporting detailed usage via real-time CDRs.

1 Telephony+ Please!

Q Why are SmartPhones so Smart with everything, except in being a Phone? Mobility is better than before, but the experience/usefulness is still only AM-radio quality, far from the telepresence capability of smartphone cameras and screens and available network bandwidth.

Q Why are the enterprise UC, Skype, GoogleTalk and all the others, providing more than plain telephony, in their own islands, in best case communicating via the POTS?

Q Why is FAX failing more and more?

Q Why are telcos concerned about losing their income to Skype and the like instead of offering something better and more attractive?

Q Why does telepresence calling require building a NEW network (as planned by OVCC to meet the high interest and obvious benefits)?

Q Why does my friend call me on my plain telephone number to ask me to start my Skype client (to use island)?

Q Why do the high capacity 4G wireless networks, with IP in the bottom, only offer VoLTE (POTS) in addition to ordinary Internet, and later maybe ViLTE? (Video over LTE)?

Q Why don't our fixed phones and IP phones have the SMS service?

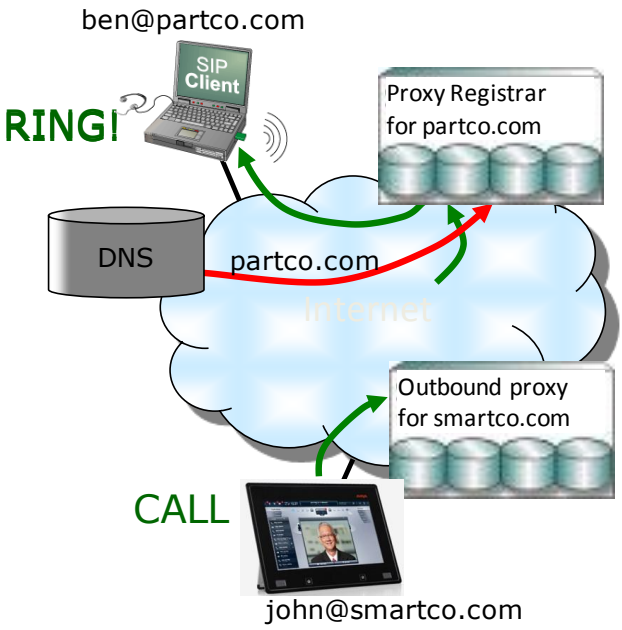
Q Why do so many interoperability issues have to be resolved just to initiate SIP trunking (connecting IP PBXs to operators' VoIP infrastructure)?

Q Why is interfacing to real-time cloud services such a hassle, with lab certifications of various elements (instead of just using a standard)?

A Network service providers have implemented VoIP as an overlay of softswitches and SBCs (IMS or SDN), where (old time) telephony is handled separately from other IP traffic, inhibiting better real-time communication and the benefits of a global IP transport network like the Internet. And the phone number connections are kept within each service provider.

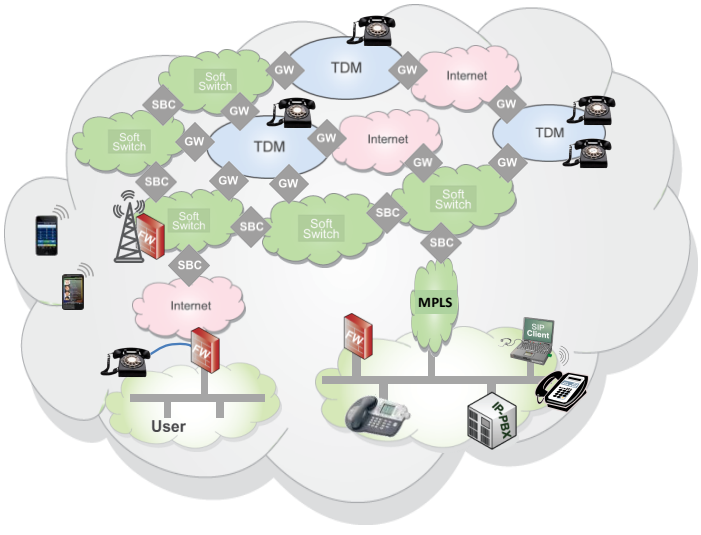
2 Time to Use SIP as Intended

SIP is a self-routing protocol. It uses DNS to find the SIP server for a called party, which knows the user's current location and forwards the call to that user. The SIP protocol was designed to take advantage of the Internet idea of a global IP transport network, where everyone reaches each other for any usage. The Internet model has proven to be tremendously successful and has changed society forever.



3 The Overlay Structure – SDN or IMS

Contrary to the SIP model, a softswitch in a network instead delivers the call to another telco/ carrier, which may be closer to the called party. They thereafter bill each other and the users as in the old telephony network. Each telco/carrier usually transports the telephone calls over its own IP network and most often hands the call over to another telco/carrier using old TDM interfaces and charging models models via an NNI (Network to Network Interface). Such structure degrades call quality, introduces interoperability issues and will not carry any service beyond POTS.



Session Delivery Network (SDN)

Even the 160 year old FAX service has now been shown to fail over SDN networks[1].

IMS networks had the ambition of global multimedia delivery, but even if IMS gets its huge and complicated standard extensions together *and* get them implemented, it will still be impossible to even agree on a price list for all the real-time applications that should appear. Application delivery should not be built into the network! After more than five years of IMS deployments, there have yet not been any multimedia calls across telco/carrier domains. IMS-specified peering is not even used between mobile phone networks, where IMS has its roots.

With the Internet+ model, every imaginable real-time application would directly be available for use between users and services. Using the SIP standard, service providers can offer flexible price plans for both old services such as POTS, SMS (text messaging) and for new services. Other usage of high quality IP delivery can simply be charged per MB, time, or flat rate where high quality is billed differently from best effort traffic.

4 Structured Service Provider Model

For telephony-type services, the network provider is often also the telephony service provider, but not always. Over IP access, there are ITSPs providing the regulated telephony service over the Internet or over private IP, e.g. via MPLS or other VPN. In addition, there are the Internet telephony islands like Skype, GoogleTalk and others, characterized by that they don't use E.164 telephone numbers natively and connect via gateways into the regulated telephony world.

In an Internet+ model;

the Network Provider (NP) provides access to the global Internet and the higher quality paths,

an ACcess Server Provider (ACSP) delivers Internet+ access via the TOQ router to the user's domain - a LAN or a smartphone - and uses the metering function to allow transport-based, service-based or session-based charging for higher quality usage,

an Internet Telephony Service Provider (ITSP) can provide the Telephony+ by providing telephone number, SIP address, and by enabling number translation (e.g. ENUM) and providing integrated access to the POTS world.

These provider functions can and will of course be combined in various ways.

With the current hardware overlay of central softswitches and SBCs, these functions get intermixed and we are stuck in a POTS-replicating world, with interoperability problems and degenerated quality, hindering the modern real-time communication that our terminals are capable of and the Internet has the bandwidth for.

Notice that critical real-time applications only will occupy a small portion of the ever-increasing available bandwidth. – The required prioritization is just a better sharing of available bandwidth.

5 Delivery to the Users

The delivery of real-time communication (via SIP and possibly also RTSP and WebRTC for other applications) all the way to the users is essential for access and interoperability. The TOQrouter provides this function, via its SIP proxy-based NAT/firewall traversal function (like a capable E-SBC), so that the SIP interface is complete and the same wherever used, at a LAN, by a smartphone, by a service in the cloud as well as for a remote user of a LAN-based SIP PBX.

Today, there are various methods for getting global SIP communication to the LAN users, however most often not provided generally, but only for a specific service, for a federation, or for a cloud service and too often with its own SIP tweaks or in a walled environment, not enabling general SIP connectivity.

E-SBCs or workaround methods are used for getting SIP communication into a private LAN. Workaround methods using keep-alive packets and relying on symmetric RTP are limiting functionality and their success depends on the firewall they have to traverse and can usually not enable higher quality access to be used.

In wireless networks the keep-alive packets also drain the batteries of smartphones, making them unattractive for use over the current Internet channel for real-time applications.

[1] Mike Coffee, CEO of Commetrex: Work in progress by SIP Forum's FoIP Task Group i3 Forum.

6 The Purpose of SDN and IMS

The functions that the SDN, but not current transport network like the Internet, may achieve are: (i) service providers can bill for their services, (ii) the ability to use a higher quality IP transport network, rather than the best effort transport offered by the public Internet, (iii) the ability to only allow trusted users - that is, subscribers to a service provider - to participate in the communication, (iv) fulfilling lawful intercept requirement and (v) fulfilling emergency calling requirements.

The Internet+ model can provide the above functions in a much better way, without using an overlay structure of softswitches and SBCs, while maintaining the reliability, scalability and good performance of a global IP transport network (all of which is destroyed by an overlay network through the introduction of massive central elements and multiple conversions).

7 The Internet+ Model

The global IP transport is simply the Internet with added quality paths. This can be provided by turning on diffserv in the routers of the Internet, or providing the higher quality on additional paths. The network providers can select different ways to provide the quality path, as long as white IP addresses are used and the quality path is routed to the Internet.

The quality path must of course be IP-peered between network providers, just like the Internet itself. The same settlement arrangement between network providers (usually settlement-free) should also work for the quality paths.

The TOQrouter can have several WAN interfaces, but still present a single Internet+ access to a user's private LAN or a smartphone on a wireless network.

8 The TOQrouter – a Smart E-SBC

Network providers almost always provide a terminating box at the end of the delivered line. It may be an MPLS router, a T1 router, a cable modem, a DSL modem/router (residential gateway) etc. A TOQrouter can be integrated in or complement such CPEs.

The Intertex IX78 E-SBC has TOQrouter capabilities in an ADSL modem also allowing VLAN ethernet access handling 100 Mbps or up to 50 voice calls. The Ingate SIParator® line of E-SBCs can handle Gbits of data and thousands of calls in TOQrouter applications. They can also encrypt the media streams over the Internet+ transport.

The TOQrouter must include a full RFC 3261 compliant SIP proxy, which routes SIP calls and messages and also translates phone numbers to SIP addresses using public or private ENUM.

Calls to PSTN devices will be routed to a PSTN gateway close to the called party or into an existing overlay network for PSTN connectivity. SIP Connect 1.1 entry points can directly be used.

The TOQrouter handles the NAT/firewall traversal and provides the same SIP interface to users and servers on a private domain as if they were connected on white IP addresses on the public side. This is highly important to allow all type of real-time applications and to avoid introducing incompatibility.

The SIP proxy in the TOQrouter is also important for its NAT/firewall traversal function and for the routing of calls and messages. The SIP proxy is also essential for the classification for prioritization of real-time traffic and for the metering and CDR generation for billing of beyond best effort Internet usage.

The TOQrouter authenticates users, or simply allows anyone on the private domain to use SIP services.

The TOQrouter is trusted to communicate with the SIP network by using Mutual TLS signaling while the service provider supplies certificates signed by a common root CA for all telephony service providers.

The access service provider can use the same certificate for a secure and authenticated connection to its management system. The secure and highly scalable TR-069 management protocol can be used for both management and CDR delivery, which will end up in an SQL database for further processing. The presence of the TOQrouter, its identity and the metering function can also be checked and authenticated via the management system.

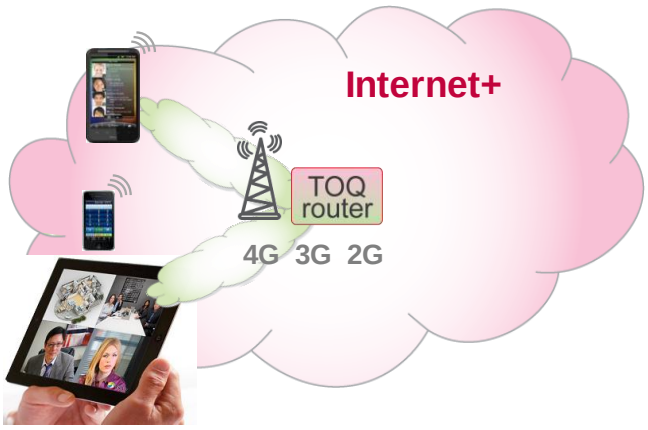
9 Architectural Benefits

The proposed architecture, without massive central elements in the communication path and

layered on top of the distributed IP layer, has good (infinite) scaling capability.

Services such as DNS and ENUM, as well as management systems based on TR-069, have their own solutions for reliability and infinite scaling.

The Internet+ model also applies to wireless networks using smartphones. In such networks, the TOQrouter router can simply be introduced at the point where the firewall for the Internet access channel for the smartphones sits, in the current 2G, 3G and 4G networks. Wireless providers will house the TOQrouter in their own premises, allowing existing management methods and CDR delivery to be used.

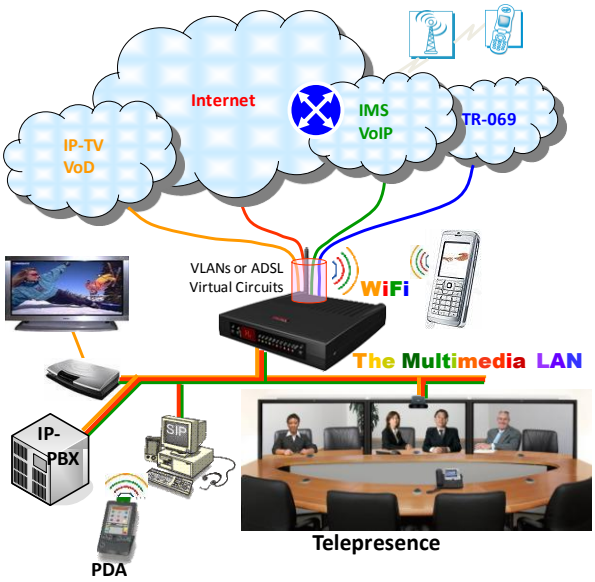


With a quality IP channel and Telephony+ capability provided by the network, mobile apps and telepresence calling utilizing such possibilities can be expected to explode. Mobile operators can charge for valuable priority bandwidth instead of providing all Internet bandwidth as flat rate best effort.

10 Only Small Extras for the Internet+

Most of the TOQ router functionality is found in competent E-SBCs (Enterprise Session Border Controllers).

These are already deployed by service providers in volume, but currently only used for SIP trunking of PBXs. In such cases, the VoIP connection usually comes on a higher quality IP pipe. This is also the case in triple-play deployments, for mass residential usage, but with pipes often capable of telepresence calling, yet only used for voice.



In the above deployment, the Intertex IX78 E-SBC is used for SIP trunking, but is actually capable of TOQrouter functions. This major European telco has a high quality VoIP network using white addresses and is routed to the Internet. An Internet+ model would here simply mean IP peering their VoIP IP network to other service providers' high quality networks, supplying an ENUM database and relying on CDRs delivered to the management system. The Intertex IX78 already provides the clean SIP interface to LAN endpoints and servers on the LAN, in parallel with its gateway approach toward the PBX and the IMS system.

Certificates and PKI are proven and used concepts. TR-069 management systems are also widely in use, especially in volume DSL deployments.

Further, Internet+ access can be deployed step by step, just seen as an extension of current networks. One provider can introduce Internet+ access for its subscribers, and the next provider then simply IP peers its quality network, etc. The billing of video calling for example could be per MB transferred or a simple flat rate for high quality usage, until e.g. per call charging is implemented.

11 Vast Gains for the Telcos

The Internet+ model will again allow telcos to earn on their core business, providing global person-to-person communication. They own the wires and can deliver quality traffic for all our benefit. And they can do this without the costly and destructive overlay of softswitches and SBCs.

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