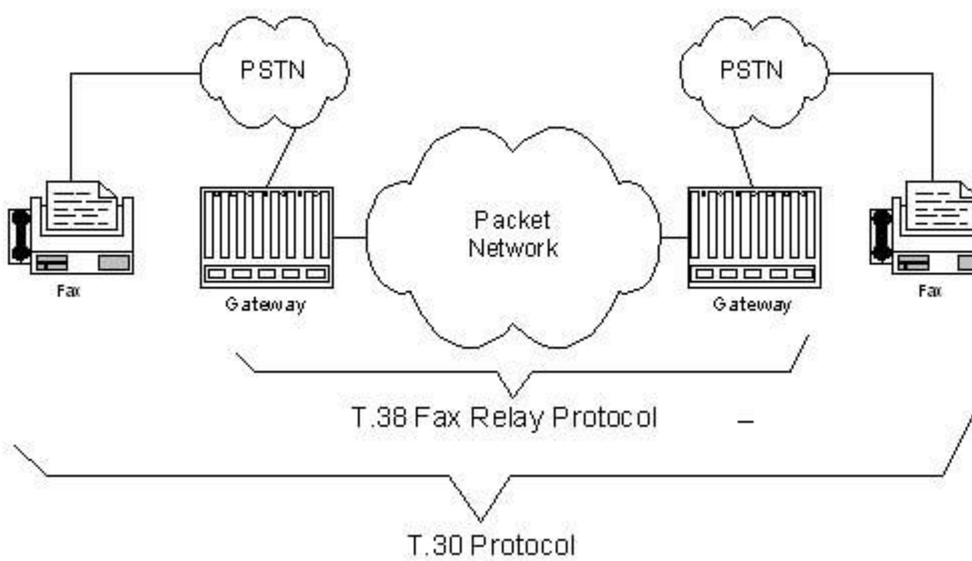
International FoIP

By Commetrex

The ITU published (or “determined,” to use ITU speak) *T.38, Procedures for Real-Time Group 3 Facsimile Communication over IP networks,* in October 1998. And it’s taken 14 years for the scope of deployment to begin to reach the international carrier industry, which is now taking up the special set of challenges of making FoIP work in international calls.

Since the amount of infrastructure involved was relatively small, initial deployments of FoIP were in enterprise IP networks (FoIP Phase I), typically as an outgrowth of VoIP deployments. Since there were no FoIP-capable carriers until the mid 2000s, enterprise FoIP stopped at the enterprise-network edge where IP-PSTN gateways converted the FoIP call to a PSTN call. Interop problems were easy to solve due to the few vendors involved.



Also, in the first half of the previous decade, the IP carriers that came out of the late-‘90s technology expansion were enlarging their footprint for Internet and VoIP services. SIP began to take over IP-telephony call control, all setting the stage for SIP trunking to be offered by Internet-telephony service providers (ITSPs). But first, the major IP carriers, such as XO Communications and Global Crossing, had to deploy T.38-capable infrastructure and offer T.38 service agreements, which they began to do just after mid-decade. Then, in 2008, the SIP Forum came out with its SIPconnect specification, spurring ITSPs to offer SIP-trunking services, handing calls off to those major IP carriers. After much trial and error, this began to work out fairly well within national borders. To be more specific, these FoIP calls worked when the SIP-trunking provider handed off to the IP carrier, which then peered with an incumbent TDM network, which delivered the fax to the destination terminal. If it got more complicated than that, additional problems arose.

But the problems that were encountered during what we call Phase II (or SIP trunking) FoIP deployment were great enough to prompt the SIP Forum to charter the SIP Forum FoIP Task Group in September 2008 to do something about the problems with Phase II deployments.

Notice that the legacy PSTN carriers were FoIP no-shows throughout the decade. But, at the turn of the decade, the approach of the day when the global carrier network would make the transition from TDM to IP began to loom large, and the i3 Forum, the carrier organization chartered to enable the carrier community to successfully make the transition, asked what had to be done to take international FoIP along with the transition. This was a timely question since it roughly coincided with the formation of the SIP Forum FoIP Task Group, prompting the two organizations to join forces to answer the question.

The unprecedented alliance was a good fit: Equipment vendors are strongly represented in the SIP Forum membership, and they know the technology as only the developers can. The i3 Forum members know the network operations as only they can. But the problems have to do with the deployment of the network equipment into those carrier networks, so the two groups had to work together to test, analyze, and solve the problems.

Testing was performed in two phases: Phase I consisted of 14 international carriers doing fax broadcasts to each other. There were many problems with the originating networks, many of which had never seen an FoIP call. There were problems with the FoIP server setup. But the biggest problems were the result of the carriers using SS-7 ISUP-based least-cost routing, peering with other international carriers, even if they were incapable of effectively handling an FoIP call. It was fire-and-forget least-cost ISUP routing as usual, and it didn’t work well.

But, what if there are inherent problems with T.38 and SIP so that a successful FoIP call was unlikely, even if the carrier was fully prepared? That question was answered by sending calls over the open Internet, both T.38 and G.711 pass-through. Within the US, faxes were perfect for both T.38 and G.711 pass-through. So, T.38 works in IP networks, but the chances of a successful international fax transaction when the carrier uses routing-as-usual was just over 50%...not good enough.

So, why is FoIP such a challenge? After all, VoIP works in international calls. The answer has to do with the strict timing requirements of T.30, the end-to-end fax protocol. T.30’s timing requirements assume that the transport is a TDM network, not IP with the timing uncertainties of a connectionless packet network. Not only that, the carrier networks assume each call is a voice call, or at least they require that the session begin as such, and re-Invite the session over to T.38 if a fax terminal answers. It turns out that the timing of that transition is critical, and maintaining it over multiple tandem connections is, to say the least, problematic.

So…what’s the answer? Intelligent FoIP routing is the answer.

Routing an FoIP call over multiple tandem connections is just too problematic to be practicable. This means carriers that want to provide reliable FoIP service today must route FoIP calls over validated FoIP routes, and, for the most part, that means over-the-top IP routes; not handing off the call to the next national carrier, but giving it to an IP carrier that can deliver it to the TDM incumbent in the destination local area. This is recommended since legacy carriers have yet to build in-country IP networks supporting FoIP, as have the international IP carriers.

But if you know something about FoIP and have been following this discussion, you may be wondering how all the routing magic is performed; after all, a carrier's routing entity assumes that a session is a voice call, not a fax call. And it's well known that for T.38 version 0 the re-Invite to T.38 is the responsibility of the off-ramp or called gateway, long after the call was routed. The answer is that the carriers must deploy a way for the on-ramp/calling/emitting gateway or server to signal that the call should be routed over a FoIP-qualified route.

It turns out that there are IETF RFCs, such as RFC 3840 and 3841, that allow a caller to express preferences about routing to the routing entity. The SIP Forum is currently working with some of the standards organizations to add support for "intelligent FoIP routing."