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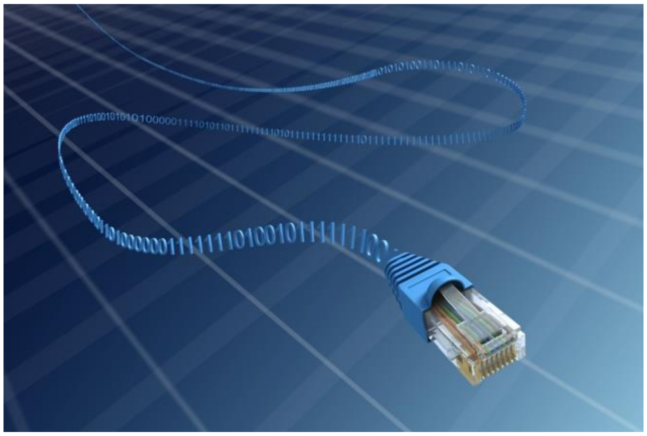

Cable IP Networks, VoIP, and the Enterprise Challenge

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Synopsis

Cable networks to date have been very successful in deploying telephone services to residential customers. In so doing the "triple play" (television, internet, and telephone) truly has been realized for this user group.

With that success the enterprise telephone PBX user should be the next logical target for the MSO. This paper will explore some of the VoIP Telecom issues they'll need to address to succeed in this potentially lucrative new market…

Cable and Telephony—the Current State of Play

MSO operators have been upgrading their networks and building out to new customers for years. The result is a network footprint that today has excellent coverage and a technology that is superior to the basic two-wire loop currently deployed by the Incumbent Local Telephone Exchange Companies (ILECS).

MSOs have done a great job of leveraging this build-out, and competing with the ILEC for basic residential telephone services. Using HFC (Hybrid Fiber Coax) technology the MSO operator has been able to duplicate the basic 2-wire residential plain old telephone services (POTs) offered by the ILECs, yet also on that same coaxial cable offer hi-speed broadband internet services.

Conversely, the telephone companies have been at a distinct disadvantage, as their two wire loop DSL technology simply cannot compete with HFC broadband IP. The speed is simply not there. That puts the MSO operator in a very interesting position, and begs the question "so what next for the cable companies?"

The ILEC's highest margin customer is its business client, a market that cable has yet to succeed in capturing on a large scale, but could. MSO networks have the bandwidth and core technology in place to aggressively compete with the ILEC for that business user. Before that takes place, some key issues must be addressed.

Cable's Next Step

Business users are very different from residential users. Organizations larger then eight employees generally use either a key system or PBX. This allows the enterprise to take advantage of trunking ratios, whereby there is a ratio of trunk circuits to the telephones that is less then one; i.e. there are fewer trunks to the phone company than there are numbers of telephone stations deployed inside the organization. This ratio varies depending on the business, however, ratios as low as 4:1 or as high as 16:1 stations per trunks are common.

In traditional telephony these trunking circuits are generally brought in via T1 circuit interfaces, and then connected to the PBX or key system. That is where the business user differs significantly from the residential user.

The MSOs for the most part cannot deliver digital TDM T1 trunks to end users. Rather, except for the larger MSO providers, to date MSOs have only succeeded in delivering two-wire POTs (plain old telephone service) circuits, intended to connect directly to a telephone station, and thus giving a station-to-trunk ratio of 1:1.

So the MSO must charge for that circuit as if was a single two-wire station telephone line, since it ties up bandwidth space on the cable net, and a station side port on its switch, regardless of whether it is being used in a call or not… just like a telco's real two-wire loop pair out on the telephone pole.

This 1:1 ratio is very unattractive to the enterprise business user. It is simply too expensive and doesn't offer the features needed by PBX users to conduct business; namely ANI (Answer Number Information) and DNIS (Dialed Number Information Services), and the ability to hunt as a trunk group into a PBX.

The DOCSIS Cable Labs T1 Solution

Cable Labs has attempted to put forth a T1/ISDN solution using HFC, and some of the larger MSO's have experimented with it. However, the technology is plagued with some serious short falls.

Problem number one is that it is expensive. T1 over DOCSIS HFC requires new expensive hardware back at the head-end office to deal with T1/TDM technology. Further, it starts putting significant pressure on the cable network itself, as it ties up bandwidth.

T1 over HFC is nothing more then a circuit of another kind, and so ties up precious cable bandwidth, regardless of whether it is being used to carry voice or not.

Probably the biggest drawback to T1 over HFC is that it is already obsolete. Enterprise users are now upgrading their PBXs to interface directly to VoIP, and specifically to SIP. A T1, as a telephone interconnect for the enterprise PBX, is rapidly becoming irrelevant. In its place, the enterprise PBX will want to take a direct RJ45 Ethernet connection and do all telephony natively via IP.

So with those two facts…

1. It is a very expensive proposition to deploy DOCSIS T1 over HFC
2. The T1 interface is already irrelevant

…it would seem unlikely that this would be a technology the cable industry would embrace on a large scale to conquer the enterprise voice market going forward.

What about Hosted VoIP?

The IP segment of the MSO network is a perfect transport medium for VoIP, and to date many MSOs have begun offering hosted VoIP services. Hosted VoIP is very similar in nature to the old telephone Centrex service, whereby the carrier creates a virtual PBX domain for the enterprise off its core end office switch.

Hosted VoIP does the exact same thing, and thus negates the need to have a PBX on location at the customer's premises. Nonetheless, there are significant issues with offering business users hosted VoIP services over cable IP networks.

First is quality of service, or QoS. Getting VoIP correct over the last mile to the end user is tricky and requires VoIP engineering if that end user plans to merge bulk enterprise data with VoIP traffic over the same cable drop. If a QoS solution is not enacted, the voice calls will be preempted by the bulk enterprise data, and give the user an unsatisfactory call experience; i.e. choppy, noisy calls.

Another major issue is traversing the corporate firewall. Hosted VoIP services must penetrate the corporate firewall. However, all firewalls are different when it comes to VoIP support, and so present immense technical and even political problems for the MSO operator to overcome.

For the most part, firewalls will require re-configuration by end user to accept VoIP traffic. In many cases, the firewalls will not even support VoIP, or the owner/administrator will have no idea how to make that reconfiguration. Corporate firewall issues present huge technical and political roadblocks for MSOs when deploying VOIP services (hosted or otherwise) to the enterprise.

Lastly, hosted VoIP simply doesn't scale. Beyond eight users hosted VoIP services lose their attraction. These products only offer 1:1 ratio between trunk and station, so in the end are expensive services that can also tie up corporate bandwidth very quickly.

In summary Hosted VoIP is not a long-term solution for the enterprise; it's simply too costly. Couple that with free PBXs (like Asterisk) running on almost free PC hardware the size of a toaster, and the enterprise will ultimately realize there is no substitute for a PBX on premises. Further, they will demand a PBX that can handle VoIP as a native signaling medium.

Then How About SIP Trunking?

The other alternative to HFC T1 and hosted VoIP technologies is "SIP-Trunking". SIP Trunking is basically a better virtual T1 over IP, with none of the channel limitations found in a T1. It can provide as many trunks to talk on as there is bandwidth to handle it.

This non-channelized nature makes SIP-Trunking a very attractive technology for the MSO operator as there are no physical hardware changes needed to handle small or large customers. All the MSO operator needs do is to tailor the bandwidth to the end user to give them the voice capacity they need.

Still, there are significant issues. Offering SIP-Trunking services to customers over cable IP networks has some considerable technical challenges that must be addressed.

To offer SIP-Trunking services the MSO back office must still be provisioned with switches. In the case of VoIP these are soft-switches, as opposed to the traditional class-5 end office TDM switch technology used by POT's and T1 telephony. Granted these devices are far less costly to deploy then their TDM counterparts, but they must be deployed none the less. Further, there are billing systems that must be provisioned and interfaced to this core VoIP switching infrastructure.

These back office infrastructure build outs can be daunting for a small or medium size MSO operator, especially in light of the fact that today most MSO's are simply outsourcing their current residential telephone needs to outside PSTN carriers; i.e. most MSO's today simply supply the base customer loop transport over cable, and then let a third party PSTN Telco handle all telephony needs at the backend. So this may be a major hurdle to overcome, as SIP-Trunking still requires the supplying organization to be fluent in telecommunications expertise.

Note: To date only a few of the larger MSO's have even experimented with this sort of a SIP infrastructure build out, as it's still a costly, and complex undertaking.

Perhaps the biggest obstacle to SIP-Trunking deployment by the cable industry lies at the customer premises. Just like hosted services SIP-Trunking still has the problems of…

- **Corporate Firewalls—**all firewalls are different, are usually difficult to setup to handle VoIP, may not even support VoIP, and/or its administrator may have no idea how to set it up for VoIP.
- **QoS—QoS** issues only become worse with the higher call volumes that SIP-Trunking services invite.
- **Security—**when offering SIP-Trunking services you must now protect at IP Layer-5 to assure that hackers and thieves do not hi-jack the service, and run up fraudulent bills on a customer's account.

Enterprise Cable VoIP Solutions

A simple solution to the corporate firewall and QoS issue is to avoid them all together by provisioning a dedicated IP cable drop just to service VoIP. That eliminates any contention between the enterprise bulk data, and eliminates the corporate firewall issues. It does nothing however to resolve the security risks.

Separated dedicated drops for VoIP are unfortunately a costly solution, and one that small and medium sized clients will likely reject. Not merging services over a single drop defeats one of the core economic reasons for moving to VoIP technology in the first place; namely that of converging communications networks down to one network to handle all media.

So what to do? To meet the challenges of QoS, dissimilar corporate firewalls, and security issues, all working on a merged VoIP/bulk-IP drop, an "Exo-Firewall solution" (EFS) is needed at the customer's premises.

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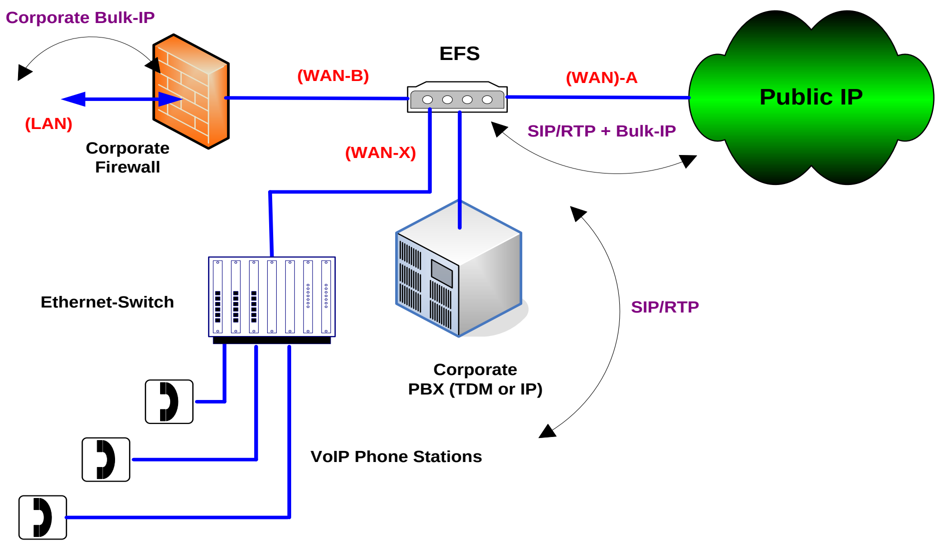

Fig 1: Exo-Firewall QoS/Security VoIP Solution

The EFS solution would ideally be a router positioned between the MSO's IP cable modem and the corporate firewall. Its attributes would be as follows:

- **EFS in Bridged Mode—**the router would be deployed in a layer-3 bridged mode, and so be unseen by the corporate firewall. In so doing the corporate firewall would have no need for re-configuration or change to accept VoIP SIP-trunking on the same IP drop

- **EFS QoS Solution—**the EFS would mitigate and manage RTP voice streams and bulk corporate data such that real-time VoIP is always given priority over all other traffic.

- **Security at Layer-5—**the EFS must offer security at IP-Layer5, and be "SIP aware". At a minimum it must only authentication and allow traffic to flow to and from the MSO carriers SIP switches, and disallow call signaling and traffic to anything outside that established SIP domain

- **EFS Must Segregate VoIP traffic—**Ideally, the VoIP office stations (VoIP telephones) should be segregated onto their own IP LAN segment, and separated from enterprise LAN traffic. This is needed both for QoS concerns, as well as for security concerns. The EFS would be the mechanism to tie this private VoIP LAN domain back to the main public WAN via the shared IP drop.

In so doing, VoIP traffic would never enter the enterprise LAN, and so never present a potential security threat. This is critical. The MSO must avoid at all costs getting involved in any way with the enterprise LAN infrastructure behind the corporate firewall, both for political as well as technical reasons.

SIP Trunking the Clear Choice for MSOs

If the major issues of QoS, disparate firewall interfaces, and VoIP security can be resolved at the customer demark, SIP-Trunking will clearly be the technology of choice for the MSO operator, and the MSO enterprise customer.

Some of the SIP-Trunking advantages for the MSO operator are…

- **Path to the Future—**With VoIP and SIP-Trunking, expensive, antiquated circuit-based T1 equipment is not needed at customers' premises, or MSO head-end. This radically lowers costs for deployment, and provides an excellent migration path for the future.

- **No Bandwidth Waste—**Bandwidth over cable is better utilized with VoIP than with HFC voice. VoIP only ties up bandwidth when it's used. This has significant (positive) ramifications as more cable customers switch over to the MSO for telephone services.

- **CPE Deployment Simplified—**SIP-Trunking allows the MSO to easily handle any number of SIP-Trunks at the customer's premises, up to the limits of the bandwidth available. This radically simplifies and streamlines installations.

- **VoIP handles the Legacy PBX—**SIP-Trunking can easily handle today's legacy TDM PBX with SIP-to-T1 or SIP-to-analog conversion gateways.

- **SIP Trunking is Ready for The Next PBX Wave—**SIP Trunking instantly readies the MSO for native SIP/VoIP interfaces, such as found on modern PBXs like Asterisk™.

SIP Trunking Also the Clear Choice for the Enterprise

For the enterprise user the advantages of SIP-Trunking are again numerous…

- **Total DID Freedom (Global Number Portability/Global Number Services)—** with VoIP the end user is freed from the PSTN rate center, and so can have any DID number terminate to any CPE site, irregardless of where that customer site sits geographically.
- **Low cost Off Premises Extensions—**by leveraging the Internet, VoIP allows off-premised extensions to remote employees at almost no cost.
- **Low cost Multi site connectivity—**SIP-Trunking allows the user to leverage the Internet and totally bypass the PSTN for communications between multi site organizations, radically lowering communications costs between sites.
- **Voice Redundancy—**with VoIP's ability to allow you to ring DID's into any location one can realize redundant enterprise VoIP services. Two ingress points of presence can be established into the enterprise. The incoming calls can then be load-shared across those two or more points, resulting in voice redundancy for the enterprise. This is almost impossible to do with traditional PSTN technologies due to cost and PSTN rate center limitations.
- **True multi-Site Enterprise Unification—**VoIP communications integrates seamlessly with other IP services and IP networks, allowing true unified communications possibilities for the enterprise
- **Converging Networks and Technologies—**converging multiple networks with technologies down to a single network and a single underlying technology offers significant cost savings possibilities.
- **Cost savings in support personnel—**multiple support groups can be collapsed down to a single group to handle all data and voice needs. The cost savings realized from that sort of merger can be considerable.
- **Cost savings in monthly recurring circuit—**IP communications from the enterprise to the outside world is generally done over IP drops that are for the most part under-utilized. By moving voice servers to that resource, and removing the stranded costs of TDM T1 circuits, more significant savings are at hand.
- **Cost savings on usage—**VoIP carriers in general bypass the ILECs, and go direct to wholesale national carries for national and international destinations.

That results in radically lower usage costs, which in turn get passed through to the end user.

Summary

It appears clear that IP networks will be the dominant transport means for most media types including voice for the foreseeable future. The reasons for this are clear; IP networks are easer to deploy, less costly to manage, and offer far more flexibility in how they handle and transport various media types.

Although VoIP offers some unique challenges due to its real-time nature, these are solvable problems, as outlined above. Therefore IP, being ready for voice transport, puts the MSO operator in an interesting position.

Will Cable Labs sometime in the near future offer a standard SIP-Trunking solution that will allow the cable industry to move forward? Probably not! So far they've had a pretty clean miss. Focusing on larger capacity circuit technologies over HFC (DOCISIS T1) is not the solution.

Rather, SIP-trunking over IP is the solution, if the three key technical issues addressed in this paper are attended to; i.e. namely…

- VoIP bandwidth allocation on the last mile
- VoIP CPE network interface (the Firewall Problem)
- VoIP security for, and VoIP separation from, the Corporate LAN

If these challenges are resolved, and the MSO figures out how to market the service, cable IP networks stand a very good chance of becoming the dominant transport of choice for small- and medium-sized enterprise voice users as the future unfolds.

If you'd like to explore Cyclix SIP-Trunking services and solutions further, please contact Cyclix sales today at:

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