IS IMS THE ANSWER?

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Abstract

Is IMS the answer? The 3GPP IP Multimedia Subsystem is a topic of hot debate among technologists in the MSO community as to its validity as a service integration platform and core technology. It is regarded by many as a solution looking for a problem and by others as a panacea for simplifying the rapid introduction of new service types that have voice as a key component. This paper discusses the real world learning garnered by Cox Communications during our technology research and prototype development beginning in 2006 and throughout 2007. Specifically addressed will be the strengths and weaknesses of Session Initiation Protocol as an enabling integration technology and the challenges of providing next generation voice services in a world where the rules of the Public Switched Telephony Network still define much of what can (and can't) be done with new voice services.

Communications Technology has been on an evolutionary path to convergence since the need to transmit computer data from one computer to another arose in the mid twentieth century. The telephone network was adapted to support transmission of data. At the same time pure data network technology evolved and over time. With the advent of internet technology it was recognized that voice service could be considered as just another data type and in many ways could be transmitted within a data network as effectively as any other data type and Voice over IP was born.

Voice over IP technology has begun to replace traditional Public Switched Telephone Network elements across the globe. In most successful cases this transformation has gone un-noticed by the end user. This apparently seamless evolution has transpired because the design of VoIP technology has followed a path of replication of telephone service to a handset. In the future, the traditional telephone handset will remain part of the Voice service network but it will not be the only interface as it has been for over 100 years. The voice interface of the future may be a video screen, mobile PDA, a utility within a web page, or possibly something that is difficult to imagine today. IP Multimedia Subsystem is a technology that lends itself to integrating Voice to other application types. But is it the best answer for how to do this? This discussion analyzes the known requirements for voice services and how IMS addresses those.

IS IMS THE ANSWER?

If IMS is the answer, what's the question? Its simple and it has nothing to with feature abstraction, common network core, service ubiquity, or any of the flashy promises we have all heard much about. There are many ways to achieve service enrichment goals and just as many advocates and pundits about the right way to do it. So if the question is re-phrased a bit to, "What does IMS do better than any other possible service architecture?", the answer is "Take care of the guy on the other end of the line while all this neat multimedia feature stuff is going on in my network for my subs."

From an architectural perspective, simplification of the call handling must be achieved by minimizing the number of times that call control must be shifted to different applications. In the legacy telephony world the Advanced Intelligent Network service invocation mechanisms that allow applications to manage call state are able to work flawlessly because the rules are very well defined and rigidly inflexible. Unfortunately those same rules limit the communication types supported to standard voice user scenarios and are not extensible to other media types or session type descriptors such as presence based routing policies and lack effective web integration capability. Voice over IP service invocation and call control based on Session Initiation Protocol lacks the rigor of AIN and consequently shifting call control makes things complex. SIP is a simple and very flexible protocol. It is flexible almost to a fault and the specifications are often interpreted differently by vendors. That's one of the biggest reasons issues still arise in SIP VoIP with features that have worked for decades in the TDM world.

Issue	**Legacy Solution**	**Interim Solution**	**IMS Solution**	**Comment**
Advanced Voice Features	AIN or PRI based call forward	SIP Re-direct and call control handoff	SIP App Server	The IMS SIP Application Server provides features, feature interaction management, and web integration
PSTN Routing and Call integrity	SS7 and TDM Interconnection	SIP CORE using ENUM	CSCF and MGCF	An effective SIP CLASS 4 Tandem easily evolves to MGCF and can provide some CSCF functions. MGCF maintains call state with the PSTN and masks the multimedia functions occurring in the IP domain.
Fixed Mobile Convergence	Dedicated FMC call control agent to move a call from one phone number to another	SIP "Call pull" via re-direct using Application Server	Presence and user preferences tell the network how to connect with the user	In IMS the association of a specific phone number to each voice endpoint isn't required.

Table 1

An ideal scenario for an operator is to map an inbound call to SIP only one time when it has entered their VoIP core and anchor its relationship to the PSTN is managed there. This activity is referred to in IMS terms as Media Gateway Control Function (MGCF).

At Cox we have realized that this approach minimizes the possibility of state mismatches between a Cox customer and the far end PSTN switch that can result in failed or dropped calls. The reality we observe today is that current generation application environments and systems want to handle service requests by taking over control of the call.

A way to visualize this general issue is think about having tried to manually set up a three-way call and then disengage yourself to allow the other two people talk. These often fail because you end up with a different count of call setups versus lines in use. The PSTN switches expect those counts to match so one will initiate a teardown. We have been looking for a way to replicate a trunk release in VoIP the same way a TDM PRI does it for years to no avail because the VoIP protocols are unable how to tell the far end its okay for the counts not to match the way ISDN does this with a channel transfer. This is one of the basic problems of PSTN replication with VoIP.

Cox encountered this issue when executing VoIP interoperability with a Directory Assistance provider at the VoIP level (SIP – SIP) for directory assistance call placement by making it a three way call with a dormant leg since the softswitches involved can't agree how to release it. We pay a direct price there using SIP ports that don't do anything but that's not the real issue. The real problem is this heavy handed approach breaks the general assumptions about call state and subsequent feature invocation becomes clumsy at best, because there is this third call leg involved that started the whole thing which, from a subsequent call treatment point of view, has no business being there. That is the general workaround for releasing a trunk today with VoIP. You don't release.

Simple features become complex because a call leg exists that shouldn't be there.

Fixed Mobile Convergence today is a form of trunk releasing that encounters this same general issue of trunk releasing compromise. This isn't to say that call control based FMC systems don't work, rather, that by not being able to execute a release with a base protocol, it is a heavy handed way to do it that results in making things other than just moving the call from one endpoint to the other incredibly complex. A subsequent consequence of employing a standalone FMC system is that eventually you end up having to host feature applications on this system because it was never designed to support services from another application environment. Providing feature transparency between mobile and fixed endpoints on your existing messaging system systems (or other apps) is where operators will struggle with integration to FMC because up to three call agents(Class 5, Tandem, and FMC)are involved for feature delivery and call control.

Subsequent expansion of the feature capabilities on one or the other platform is now required and portends painful integration until these systems have been augmented past their original design capability. The ultimate consequence is that the operator ends up with another service platform that doesn't perform well but is costly to replace because of all the investment made to integrate to full functionality needed. All this in the name of taking care of the guy on the other end of the line. The issues that IMS technology directly addresses with respect to the PSTN are listed in *Table 1*

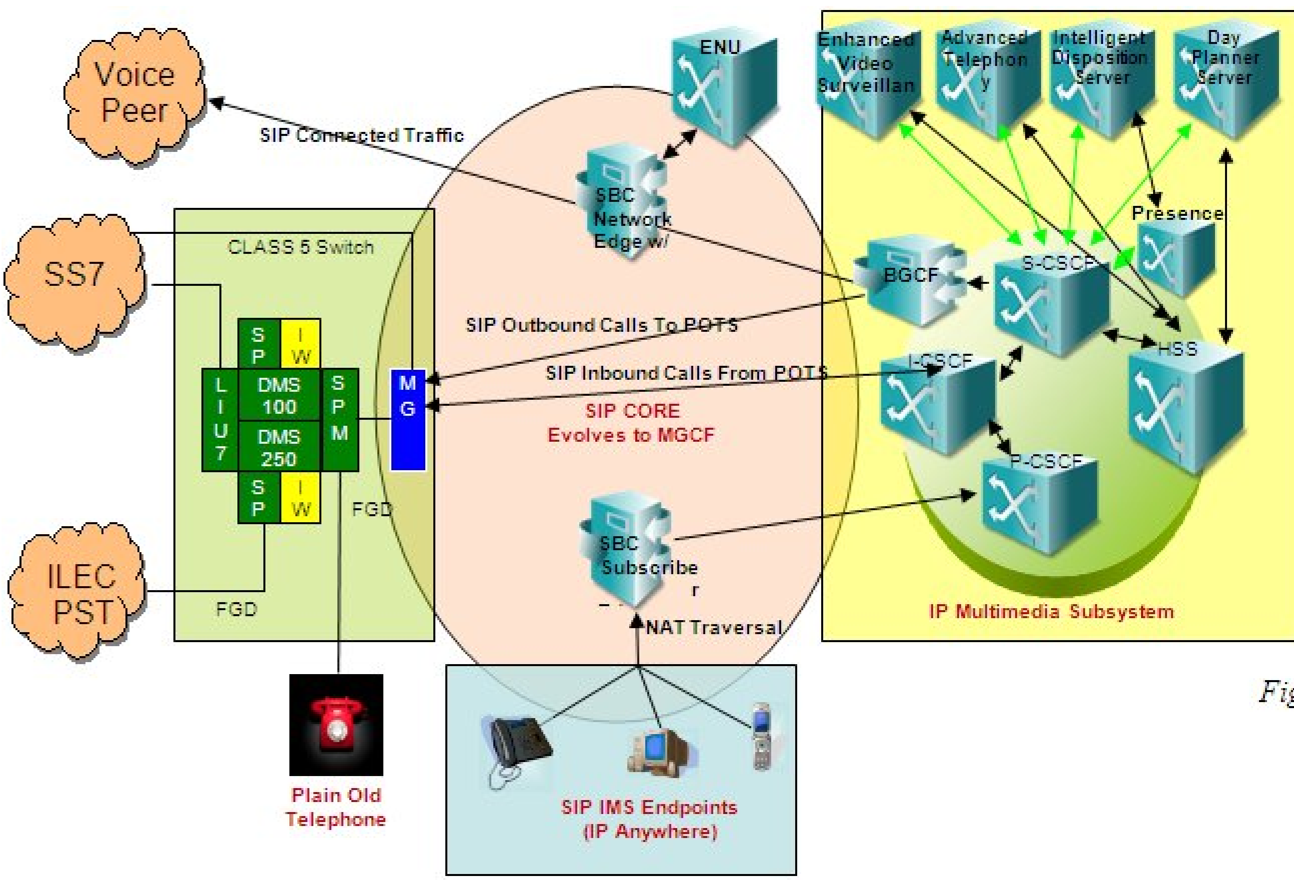

Figure 1

IMS however; addresses this directly by not allowing any one application to dominate call control with respect to the PSTN. It doesn't matter because IMS routes the inbound PSTN call to a subscriber(fully qualified domain name), not an endpoint(phone number).*Fig 1*

The concept of the far end PSTN switch that has to abide by PSTN rules doesn't exist inside of IMS, only at its border(MGCF). A good MGCF looks like just another route for a service to IMS applications. Many of today's available "stovepipe" application systems promising IMS compliance in the future but this will be a real challenge for them because when you closely inspect virtually any current generation VoIP application server you will probably see a device originally designed as a softswitch that tends to behave like one by handling connection state of that far end PSTN switch.

There are exceptions of course, some which rely on AIN and the TDM Class 5 switch it is attached to and some telephony application servers that function well being treated as a giant VoIP PBX until IMS liberates them totally from the PSTN. Good pre-IMS app servers at a minimum mask control of the call by staying in synch with the MGCF.

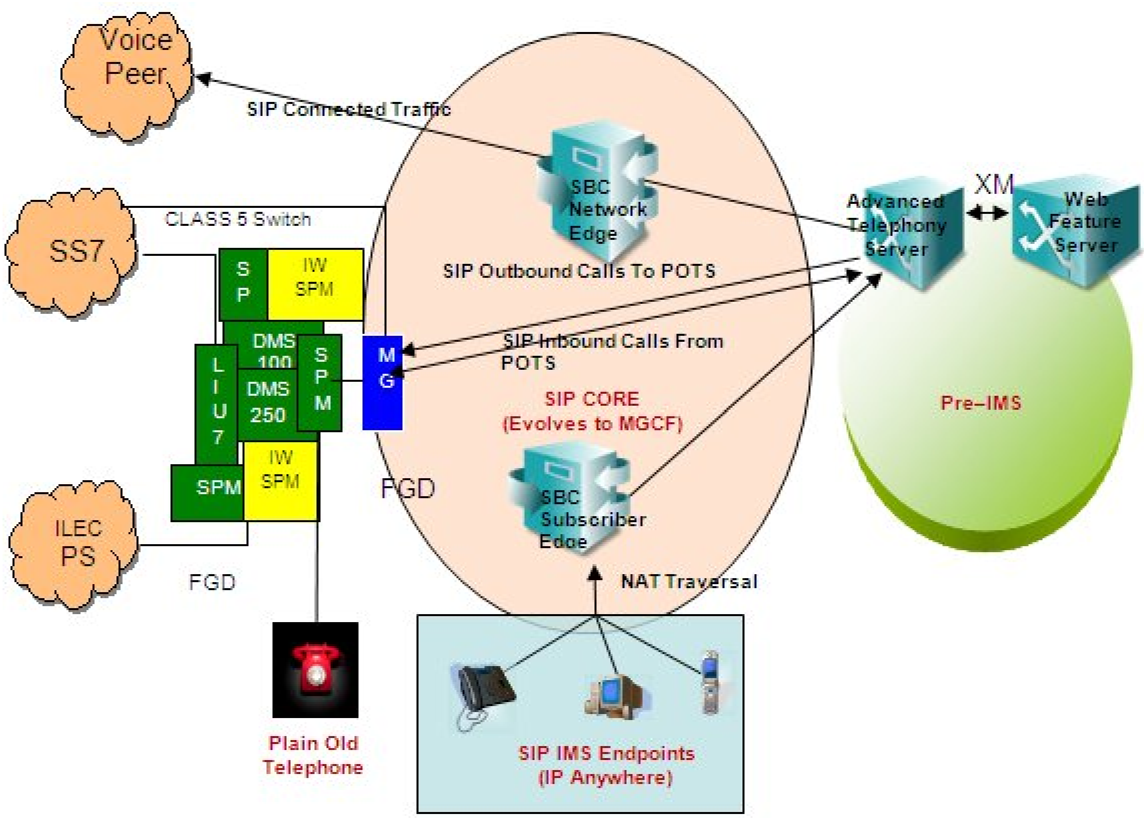

Figure 2

This opens the door to a form of FMC that can be utilized until IMS application integration has reached the level of maturity needed to extend those services to a highly reliable voice offering

This does limit the offering only to endpoints capable of supporting SIP. Inside IMS the distinction of fixed/mobile disappears so FMC based on call control handoff is meaningless. Getting applications to integrate seamlessly now becomes a design quality issue no longer restricted by the limitations imposed at the call routing layer. Feature interaction management is becoming a discipline unto itself that VoIP engineers will be practicing for as long as can be imagined.

The bottom line is that from a technology standpoint any FMC or Unified Messaging design using available application servers that directly interface to the PSTN CLASS 5 switch is a big diversion away from IMS. As such any focus on nailing up a design for FMC and UM under conventional terms becomes a full diversion away from IMS development because the core skills focused on IMS or pre IMS today are the same ones that must be used for interim solutions. Essentially a doubling of effort is required to pursue both paths.

An additional consequence is that to get to IMS, engineering teams will have to expend effort to undo the connected to legacy infrastructure adding yet more development cycles. Our experience at Cox has been that the effort to migrate systems rivals or surpasses replacing them entirely.

If you as an operator ask “Ok, what should I do instead?” the answer today is focus effort on building the MGCF that allows non-possessive application servers to work in your network and start identifying and integrating those application servers. *Fig 2* Does that get an operator to wow factor features as quickly as stovepipe systems directly connected to legacy infrastructure? Probably not but it will get you to where you really want to be faster than taking a big detour and it will allow you to stay there when you get there. Implement a uniform and reliable method of taking care of the guy on the other end of the line and there are only a few steps beyond that needed to realize the service rich environment of IMS.