International Packet Communications Consortium

Reference Architecture

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International Packet Communications Consortium

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Letter from the International Packet Communications Consortium

Dear IPCC Member:

What is a softswitch? Ask 10 people and you may get as many as 15 different answers! Which is why the International Packet Communications Consortium (IPCC) published this Reference Architecture. Defining the many functional elements that constitute a "softswitch" is an important step in promoting interoperability and clarifying the confusion that now exists among the providers and consumers of Voice over IP (VoIP) products and services.

This document is not meant to serve as an implementation guide, and its content is not intended to stifle innovation. Its purpose is actually quite benign—but vitally important: *to bring a consistent terminology to the business of next-generation packet voice networks*. In effect, the IPCC Reference Architecture is a depiction of the various aspects of VoIP networks as distinct, modular entities as the building blocks of next-generation voice networks. How the various entities are implemented is left entirely to the vendor and/or service provider.

IPCC members are encouraged to adopt the functional terms used in this Reference Architecture. Ours is not a simple business, so the architecture may appear (at first) to be fairly complicated. That's because it was crafted to be comprehensive. Yet every attempt was made to also make the architecture both intuitive and flexible. Use the portions that best fit your business. But please do use it! Once we all begin speaking the same language, the industry will be in a position to better understand the many advantages and benefits of VoIP.

The document is organized into five sections that drill down to progressively greater detail. The first section highlights the four *Functional Planes* employed in the architecture. This is followed by a somewhat detailed description of the eight key *Functional Entities* that make up an end-to-end VoIP network. The next section on *Media Gateway Controller Building Blocks* delves deeper into aspects of this all-important VoIP network element. The general discussion presented in the first three sections is followed by 10 popular *Network Examples*. The document concludes with an Appendix that lists all of the acronyms used throughout, including some references to applicable standards.

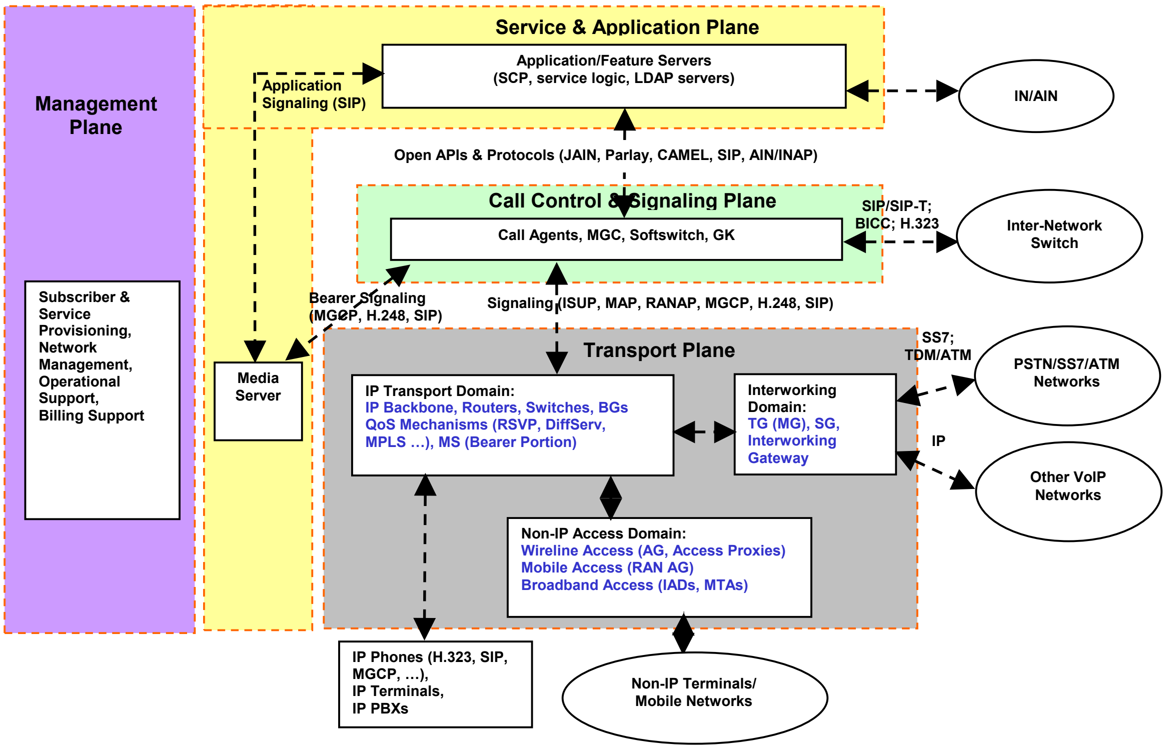
Finally, this is also a living document. Comments are both welcomed and encouraged. As the Reference Architecture evolves, the IPCC will publish updated versions.

Thank you for your participation in the International Packet Communications Consortium.

Contents

Functional Planes

The functional planes represent the broadest level of separation between the functional entities in a Voice over IP (VoIP) network. There are four distinct functional planes employed by the IPCC to describe the functioning of an end-to-end VoIP network: Transport; Call Control & Signaling; Service & Application; and Management.



Transport Plane

The transport plane is responsible for the transport of messages across the VoIP network. These messages may be for call signaling, call and media setup, or media. The underlying transport mechanism(s) for these messages may be based on any technology that satisfies the requirements for carrying these types of traffic.

The Transport Plane also provides access for signaling and media with external networks, or terminals to VoIP networks. Often the Transport Plane devices and functions are controlled by functions in the Call Control & Signaling Plane.

The transport plane is further divided into three domains: IP Transport Domain; Interworking Domain; and Non-IP Access Domain:

IP Transport Domain

The IP Transport Domain provides the transport backbone and routing/switching fabric for transporting packets across the VoIP network. Devices like routers and switches belong to this domain. Devices that provide Quality of Service (QoS) mechanisms and policies for the transport also belong to this domain.

Interworking Domain

The devices in the Interworking Domain are primarily responsible for the transformation of signaling or media received from external networks into a form that can be sent among the various entities in the VoIP network and vice versa. It consists of devices like Signaling Gateways (signaling transport conversion between different transport layers), Media Gateways (media conversion between different transport networks and/or different media), and Interworking Gateways (signaling interworking on the same transport layer but with different protocols).

Non-IP Access Domain

The Non-IP Access Domain applies primarily to non-IP terminals and wireless radio networks that access the VoIP network. It consists of Access Gateways or Residential Gateways for non-IP terminals or phones, ISDN terminals, Integrated Access Devices (IADs) for DSL networks, Cable Modem/Multimedia Terminal Adaptors (MTAs) for HFC networks, and Media Gateways for a GSM/3G mobile radio access network (RAN). Note that the IP terminals, like a SIP phone, will directly connect to the IP Transport Domain, without going through an Access Gateway.

Call Control & Signaling Plane

The Call Control & Signaling Plane controls the major elements of the VoIP network, especially in the Transport Plane. The devices and functions in this plane carry out call control based on signaling messages received from the Transport Plane, and handle establishment and teardown of media connections across the VoIP network by controlling components in the Transport Plane. The Call Control & Signaling Plane consists of devices like the Media Gateway Controller (a.k.a. Call Agent or Call Controller), Gatekeepers and LDAP servers.

Service & Application Plane

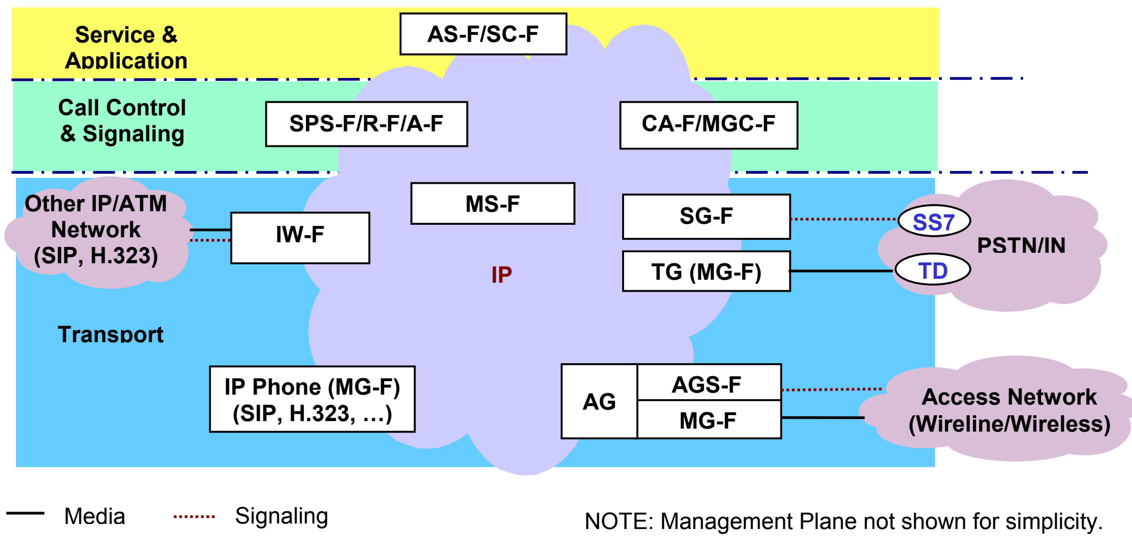
The Service & Application Plane provides the control, logic and execution of one or more services or applications in a VoIP network. The devices in this plane control the flow of a call based on the service execution logic. They achieve this by communication with devices in the Call Control & Signaling Plane. The Service & Application Plane consists of devices like Application Servers and Feature Servers. The Service & Application Plane may also control specialized bearer components, such as Media Servers, that perform functions like conferencing, IVR, tone processing, and so on.

Management Plane

The Management Plane is where functions such as subscriber and service provisioning, operational support, billing and other network management tasks are handled. The Management Plane can interact with any or all of the other three planes through industry standard (e.g. SNMP) or proprietary protocols and APIs.

Functional Entities

The functional entities are the logical entities of a VoIP network. This section describes the major functional components of the IPCC Reference Architecture. Note that these are functions and *not* physical product descriptions. The various functions may physically reside on standalone devices or in various combinations on multi-function platforms. As such, there are virtually an unlimited number of ways to bundle the various functions into physical entities.



NOTE: Management Plane not shown for simplicity.

IPCC Functions

AS-F: Application Server Function	**SC-F:** Service Control Function	**CA-F:** Call Agent Function
MGC-F: Media Gateway Controller Function	**SPS-F:** SIP Proxy Server Function	**R-F:** (Call) Routing Function
A-F: Accounting Function (e.g., AAA)	**MS-F:** Media Server Function	**SG-F:** Signaling Gateway Function
MG-F: Media Gateway Function	**IW-F:** Interworking Function	**AGS-F:** Access Gateway Signaling Function

The diagram above shows 12 different functions. Understanding the autonomy of all 12 functions is an important characteristic of the IPCC Reference Architecture.

Media Gateway Controller Function (MGC-F)
a.k.a. Call Agent or Call Controller

The MGC-F provides the call state machine for endpoints. Its primary role is to provide the call logic and call control signaling for one or more media gateways.

MGC-F Characteristics:

- Maintains call state for every call on a media gateway
- May maintain bearer states for bearer interfaces on the MG-F
- Communicates bearer messages between two MG-Fs, as well as with IP phones or terminals
- Acts as conduit for media parameter negotiation
- Originates/terminates signaling messages from endpoints, other MGC-Fs and external networks
- May interact with the AS-F for the purposes of providing a service or feature to the user

- May manage some network resources (e.g. MG-F ports, bandwidth etc.)
- May provide policy functions for endpoints
- Interfaces to R-F/A-F for call routing, authentication and accounting
- May participate in management tasks in a mobile environment (mobility management is generally part of the CA-F)
- Applicable protocols include H.248 and MGCP

Call Agent Function (CA-F) and Interworking Function (IW-F)

The CA-F and IW-F are subsets of the MGC-F.

The Call Agent Function (CA-F) exists when the MGC-F handles call control and call state maintenance. Examples of CA-F protocols and APIs include:

- SIP, SIP-T, BICC, H.323, Q.931, Q.SIG, INAP, ISUP, TCAP, BSSAP, RANAP, MAP and CAP (mobile)
- Open APIs (JAIN, Parlay, etc.)

The Interworking Function (IW-F) exists when the MGC-F performs signaling interaction between different signaling networks (e.g. SS7 and SIP). Examples of IW-F protocols include H.323/SIP and IP/ATM.

Call Routing and Accounting Functions (R-F/A-F)

The R-F provides call routing information to the MGC-F, while the A-F collects call accounting information for billing purposes. The A-F can also have a broader role embodied by the common AAA functionality of authentication, authorization and accounting in remote access networks. The primary role of both functions is to respond to requests from one or more MGC-Fs, directing the call or its accounting to terminating endpoints (other MGC-Fs) or services (AS-Fs).

R-F/A-F Characteristics:

- Provides routing function for intra- and inter-network call routing (R-F)
- Produces details of each session for billing and planning purposes (A-F)
- Provides session management and mobility management
- May learn routing information from external sources
- May interact with the AS-F for the purposes of providing a service or feature to the user
- May operate transparently to the other entities in the signaling path
- Many R-Fs and A-Fs can be chained together in a sequential or hierarchical manner
- The R-F/A-F is often integrated with the MGC-F. However, just as is the case with the AS-F, an integrated R-F/A-F/MGC-F can also request services of an external R-F/A-F.
- The A-F collects and emits per-call accounting information. The AS-F emits accounting information for enhanced services, such as conferences and premium information services.
- Applicable protocols for the R-F include ENUM and TRIP
- Applicable protocols for the A-F include RADIUS and AuC (for mobile networks)

SIP Proxy Server Function (SPS-F)

The most common embodiment of the R-F and A-F is as a SIP Proxy Server. For this reason, the IPCC recognizes a separate SIP Proxy Server Function (SPS-F).

Signaling Gateway Function (SG-F) and Access Gateway Signaling Function (AGS-F)

The SG-F provides a gateway for signaling between a VoIP network and the PSTN, whether SS7/TDM- or BICC/ATM-based. For wireless mobile networks, the SG-F also provides a gateway for signaling between an IP-based mobile core network and PLMN that is based on either SS7/TDM or BICC/ATM. The primary role of the SG-F is to encapsulate and transport PSTN (ISUP or INAP) or PLMN (MAP or CAP) signaling protocols over IP.

The AGS-F provides a gateway for signaling between a VoIP network and circuit-switched access network, whether V5- or ISDN-based. For wireless mobile networks, the AGS-F also provides a gateway for signaling between an IP-based mobile core network and PLMN that is based on either TDM or ATM. The primary role of the AGS-F is to encapsulate and transport V5 or ISDN (wireline), or BSSAP or RANAP (wireless) signaling protocols over IP.

SG-F Characteristics:

- Encapsulates and transports PSTN signaling protocols (e.g. SS7) using SIGTRAN to the MGC-F or another SG-F
- For mobile networks, encapsulates and transports PSTN/PLMN signaling protocols (e.g. SS7) using SIGTRAN to the MGC-F or another SG-F
- The interface from the SG-F to the other entities is a protocol interface when the SG-F and MGC-F or other SG-F are not co-located (e.g. SIGTRAN)
- One SG-F can serve multiple MGC-Fs
- Applicable protocols include SIGTRAN, TUA, SUA and M3UA over SCTP

AGS-F Characteristics:

- Encapsulates and transports V5 or ISDN signaling protocols (e.g. SS7) using SIGTRAN to the MGC-F
- For mobile networks, encapsulates and transports BSSAP or RANAP signaling protocols (e.g. SS7) using SIGTRAN to the MGC-F
- The interface from the AGS-F to the other entities is a protocol interface when the AGF-F and MGC-F or other AGF-F are not co-located (e.g. SIGTRAN)
- One MGC-F may serve many AGS-Fs
- Applicable protocols include SIGTRAN, M3UA, IUA and V5UA over SCTP

Application Server Function (AS-F)

The AS-F is the application execution entity. Its primary role is to provide the service logic and execution for one or more applications and/or services.

AS-F Characteristics:

- May request the MGC-F to terminate calls/sessions for certain applications (e.g. voice mail or conference bridge)
- May request the MGC-F to re-initiate call features (e.g. find me/follow me or pre-paid calling card)
- May modify media descriptions using SDP
- May control an MS-F for media handling functions
- May be linked to Web applications or have Web interfaces
- May have an API for service creation
- May have policy, billing and session log back-end interfaces
- May interface with MGC-Fs or MS-Fs
- May invoke another AS-F for additional services or to build complex, component-oriented applications
- May use the services of an MGC-F to control external resources
- Applicable protocols include SIP, MGCP, H.248, LDAP, HTTP, CPL and XML
- Applicable open APIs include JAIN and Parlay

Often the combination of the AS-F and the MGC-F provides enhanced call control services, such as network announcements, 3-way calling, call waiting and so on. Rather than connecting the AS-F and MGC-F with a protocol, vendors often use an API between the AS-F and MGC-F when they are implemented in a single system. In this embodiment, the AS-F is known as a "Feature Server."

Service Control Function (SC-F)

The Service Control Function (SC-F) exists when the AS-F controls the service logic of a function. For this reason, the IPCC recognizes a separate Service Control Function (SPS-F). Examples of SC-F protocols include INAP, CAP and MAP; open APIs include JAIN and Parlay.

Media Gateway Function: (MG-F)

The MG-F interfaces the IP network with an access endpoint or network trunk, or a collection of endpoints and/or trunks. As such, the MG-F serves as the gateway between the packet and external networks, such as the PSTN, mobile network, etc. For example, the MG-F could provide the gateway between an IP and circuit network (e.g. IP to PSTN), or between two packet networks (e.g. IP to 3G or ATM). Its primary role is to transform media from one transmission format to another, most often between circuits and packets, between ATM packets and IP packets, or between analog/ISDN circuits and packets as in a residential gateway.

MG-F Characteristics:

- Always has a master/slave relationship with the MGC- F that is achieved through a control protocol such as MGCP or MEGACO
- May perform media processing functions such as media transcoding, media packetization, echo cancellation, jitter buffer management, packet loss compensation, etc.
- May perform media insertion functions such as call progress tone generation, DTMF generation, comfort noise generation, etc.

- May perform signaling and media event detection functions such as DTMF detection, on/off-hook detection, voice activity detection, etc.
- Manages its own the media processing resources required to provide the functionality mentioned above
- May have the ability to perform digit analysis based on a map downloaded from the MGC-F
- Provides a mechanism for the MGC-F to audit the state and capabilities of the endpoints
- Is not required to maintain the call state of calls passing through the MG-F; the MG-F only maintains the connection state of the calls it supports
- A SIP phone is an MG-F and MGC-F in a single box
- A SIP-capable gateway is an MG-F and MGC-F in a single box
- "Hair pinning" of a call by the MG-F directed toward the source network may occur under control of the MGC-F
- Applicable protocols include RTP/RTCP, TDM, H.248 and MGCP

Media Server Function (MS-F)

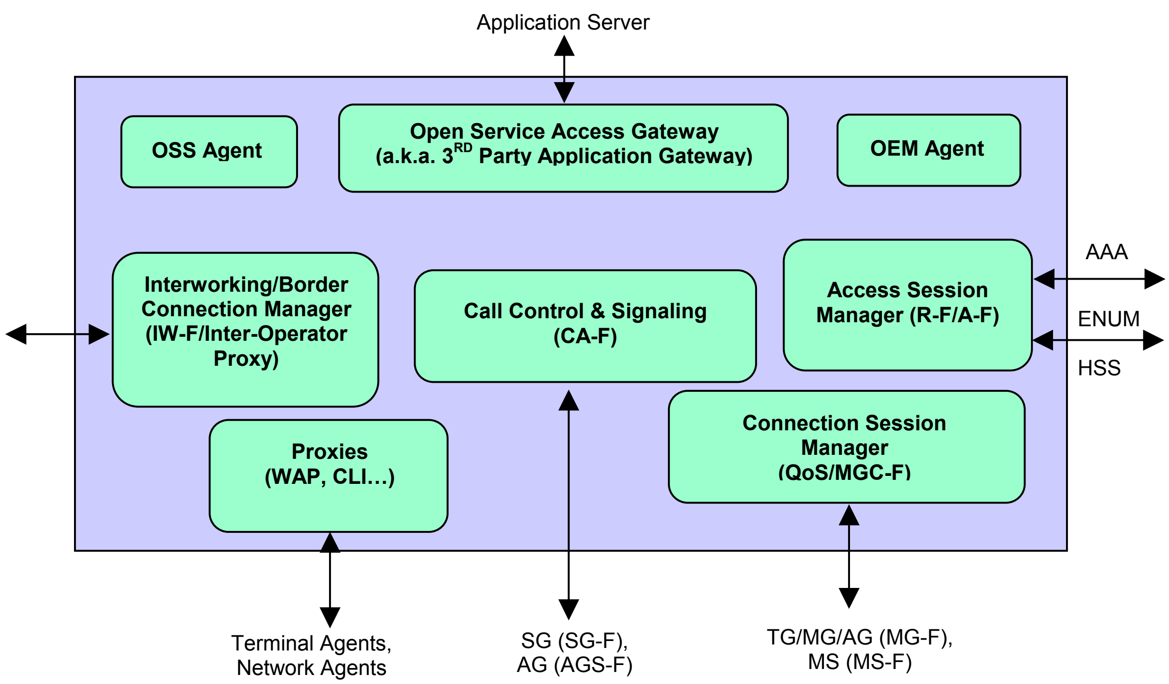
The MS-F provides media manipulation and treatment of a packetized media stream on behalf of any applications. Its primary role is to operate as a server that handles requests from the AS-F or MGC-F for performing media processing on packetized media streams.

MS-F Characteristic:

- Support for multiple codecs and transcoding
- Support for control by multiple AS-Fs or MGC-Fs
- Support for multiple concurrent capabilities:
    - digit detection
    - streaming of tones and announcements (any multimedia file)
    - algorithmic tone generation
    - recording of multimedia streams
    - speech recognition
    - speech generation from text
    - mixing (conference bridge)
    - fax processing
    - voice activity detection and loudness reporting
    - scripted combinations of the above
- Performs under the control of an AS-F or MGC-F through a control protocol, with either tight coupling (resource control) or loose coupling (function invocation or scripts)
- Applicable protocols include SIP, MGCP and H.248

Media Gateway Controller Building Blocks

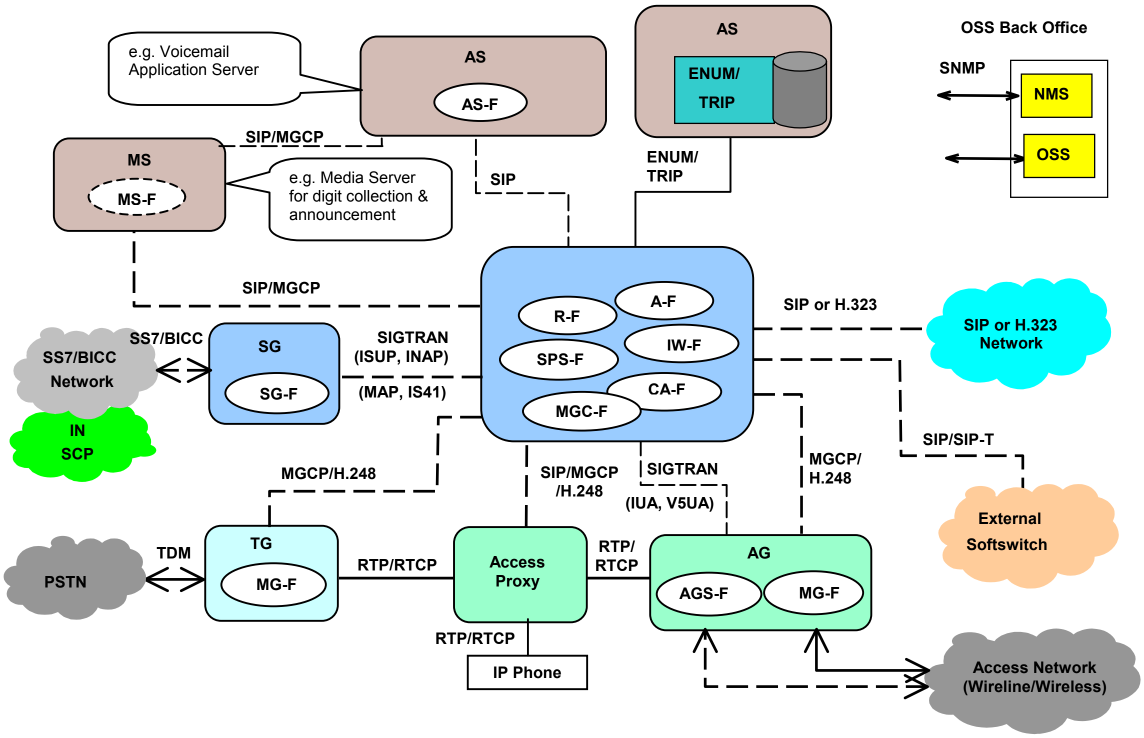
The Media Gateway Controller (MGC) is one of the key physical elements of a VoIP network. There have been many different implementations of the MGC and it is known by different names, including Softswitch, Call Agent, Call Controller and others. Shown here are just some of the many possibilities available to equipment manufacturers and service providers under the IPCC Reference Architecture.



Most "Media Gateway Controller" systems today implement other functions in addition to the MGC-F. The other functions shown here (CA-F, IW-F, R-F and A-F) could be collocated on the same physical platform or distributed across different systems that together constitute a total MGC solution. Of course, for load-balancing and availability purposes, the MGC can be implemented in a cluster of systems.

The functional building blocks in this "Media Gateway Controller" include the Connection Session Manager (MGC-F), Call Control & Signaling (CA-F), the Interworking/Border Connection Manager (IW-F) and the Access Session Manager (R-F/A-F). Other features implemented in the MGC shown include an Open Service Access Gateway, application Proxies, and OSS and OEM Agents. The Open Service Access Gateway and application Proxies together provide the discrimination and distribution of signaling and media for separate open and proprietary applications, respectively. The OSS and OEM Agents interface to external OSS/OEM managers located in an operational support center for network management, service and network provisioning, maintenance, and so on.

MGC Implementation Example



NOTE: The dashed lines represent the realization of information flows achieved by selecting specific protocol and interfaces to meet service requirements: this figure and its associated protocol choices is informative only.

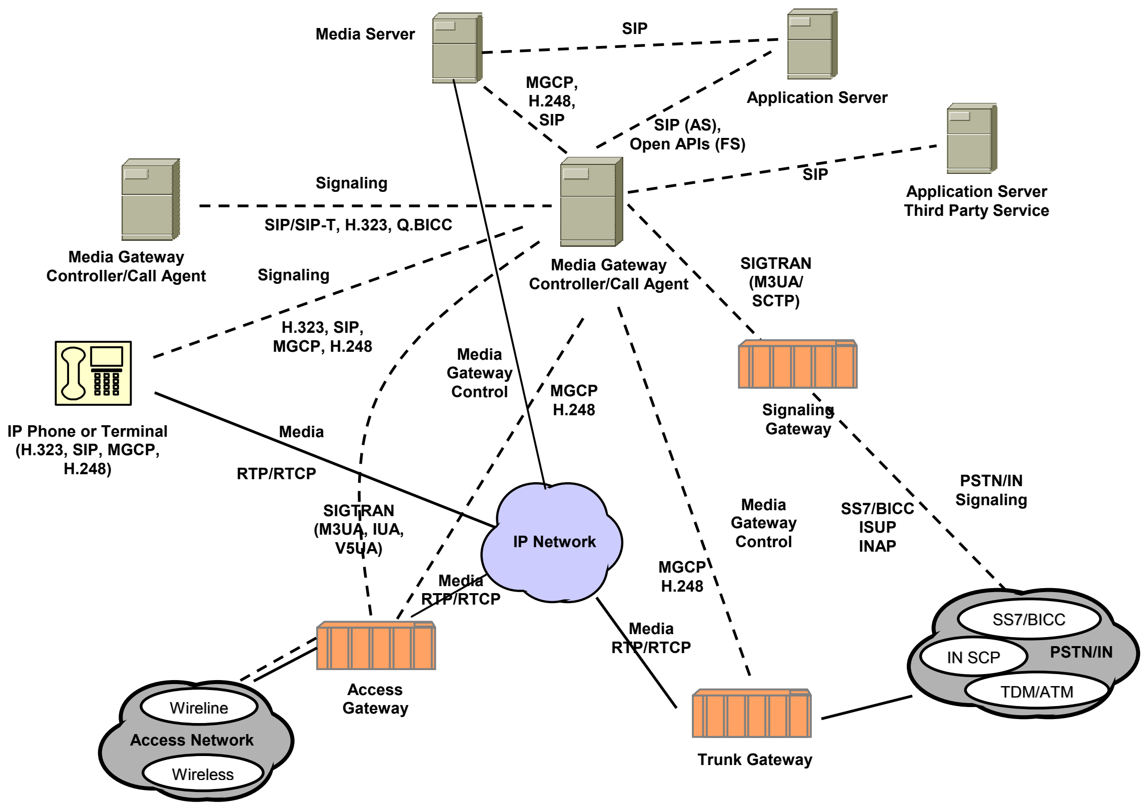
In this example implementation of an MGC, the MGC controls the Trunking Gateways, Media Gateways and Media Servers using one or more of the media gateway control protocols through its MGC-F. The MGC originates and terminates signaling messages to and from endpoints, other MGCs and external networks. Specifically, the Call Control & Signaling (CA-F) block performs this function. The Call Control & Signaling block also maintains the states of each call.

The example MGC system shown here also implements these other functions: Routing & Call Accounting (R-F/A-F), a SIP Proxy Server (SPS-F) and an Interworking/Border Connection Manager (IW-F). The MGC communicates, via the Open Service Gateway Block (or application Proxy), with the application servers for services that are not native to the MGC. This could be done through various service control protocols and APIs, such as SIP, JAIN and Parlay.

Network Examples

This section presents 10 examples of VoIP network configurations in a wide range of popular applications. Each example shows how the various functional entities might come together in physical devices and logical interactions to make up a VoIP network. In other words, each is just that—an *example* of one possible implementation, and no example is intended to represent a best or even a preferred approach.

Wireline Network



The physical entities in this example consist of the Media Gateway Controller (MGC), Applications Server (AS), Trunk Gateway (TG), Access Gateway (AG), Signaling Gateway (SG) and a Media Server (MS).

The Media Gateway Controller (MGC) in this example is carrying out the MGC-F, R-F and A-F. The MGC terminates all the signaling (either directly or transported over IP) and carries out signaling interworking (e.g. a SIP phone wanting to signal to a PSTN network). It controls both the TG or AG for allocation of media resources. The MGC also authenticates and routes calls into the VoIP network and provides accounting information. Finally, the MGC interacts with other MGCs using either SIP/SIP-T, H.323 or Q.BICC.

The Application Server (AS) has the service logic for applications, such as voice mail. Calls requiring these functions can be either handed over by the MGC to the AS for service control, or the application server can provide the information required for the service logic execution to the MGC. The AS can control the Media Server directly or pass control of the MS to the MGC.

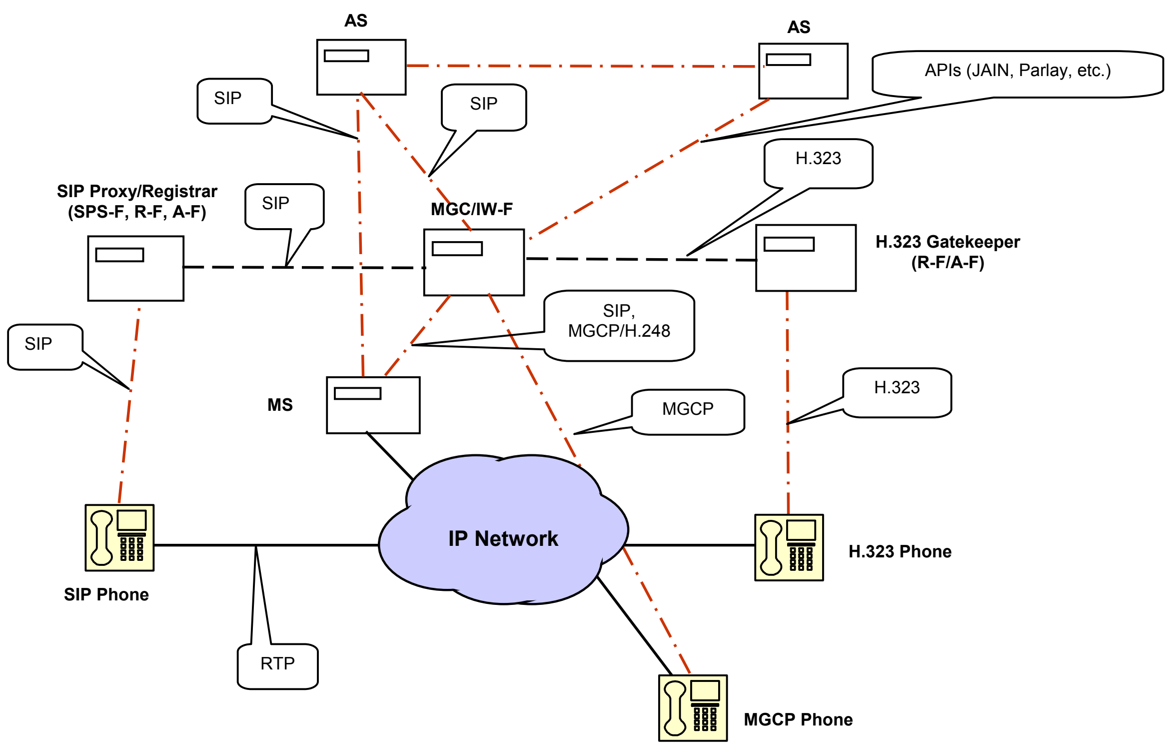
The Trunk Gateway (TG) terminates the physical media carrying the bearer (voice streams) from the PSTN, transcodes the bearer, and transports it over IP into the IP network. The TG is controlled by the MGC.

The Access Gateway (AG) serves as the interface between the IP network and any wireline or wireless access network. The AG transports signaling over IP to the MGC while transcoding the bearer and transporting it over IP to an IP endpoint or to a TG for sending it to a circuit or other packet network. The MG-F of the AG is controlled by the MGC.

The Signaling Gateway (SG) terminates the physical media for the PSTN signaling and transports the PSTN signaling to and from the MGC over IP.

The Media Server (MS) might perform tasks like announcements and digit collection, although the Access Gateway normally handles digit collection in most cases. The MS can be controlled either by the MGC, the Application Server, or both.

All-IP Network



This example shows an all-IP network with the following components (which could be grouped physically in many different ways):

- IP phones that can support SIP, MGCP and/or H.323
- H.323 Gatekeeper – domain administrator for H.323 based endpoints (R-F/A-F)
- Media Server – for providing advanced media functionality, such as IVR (MS-F)
- SIP Proxy – routing and address translation entity for SIP-based endpoints (R-F/A-F)

- Application Server – logical entity where services are hosted. One AS can also communicate with other Application Servers, which may implement services using JAIN, Parlay, etc. (AS-F)
- MGC/IW-F – In this architecture this entity is used for interworking protocols; that is, for when SIP, H.323 and MGCP endpoints that need to communicate with each other. The MGC is also used for routing and accounting functions for controlling MGCP phones.
- SIP Registrar – a SIP entity that keeps current locations for SIP devices

The SIP phones update the SIP Registrar with their current address and use the proxy for routing. In cases where a SIP phone needs to communicate with non-SIP entities, it signals the MGC, which in this example handles the IW-F with the other protocols. Note that it is also possible to offer legacy phone features to SIP phones via the AS/MGC.

The MGCP phones contact the MGC for establishing and tearing down of calls. The MGC also provides the R-F, A-F and IW-F for the MGCP phone.

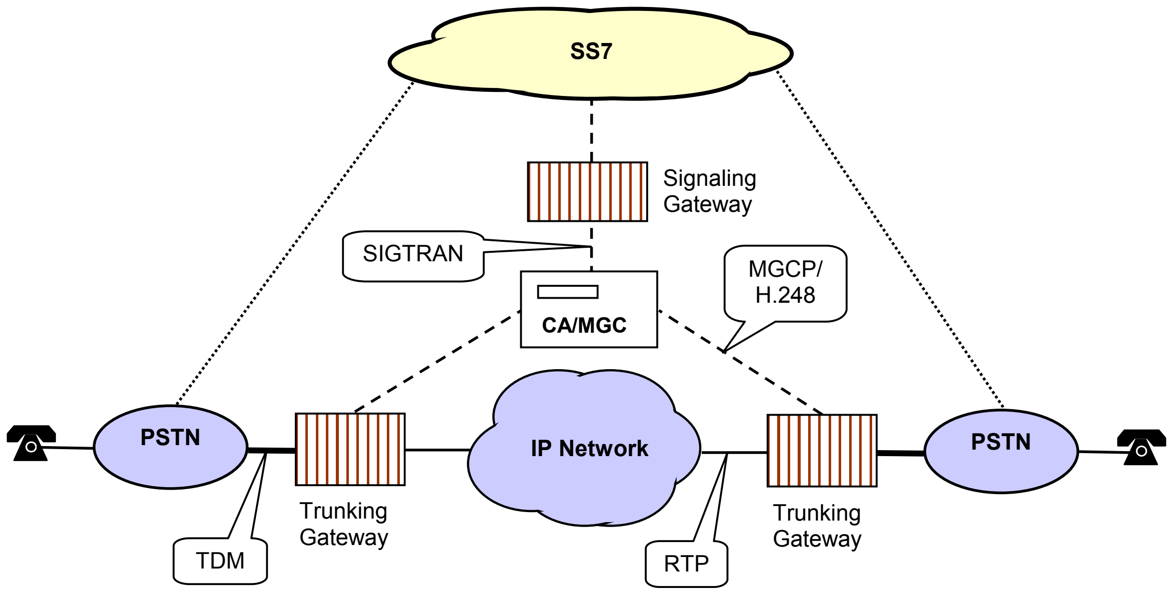
In the case of an H.323 phone, the H.323 Gatekeeper provides the R-F and A-F. The MGC provides protocol IW-F when required to communicate with non-H.323 networks.

In this case, media messages flow end-to-end between all media-capable devices using RTP/RTCP, regardless of the signaling protocol used. Hence, MGs are not required. A Media Gateway may be used if transcoding between different codecs is required, either on- or off-net.

The billing and accounting information may be exchanged between the gatekeeper, SIP proxy and/or the MGC for overall accounting purposes, especially in case of multi-protocol calls.

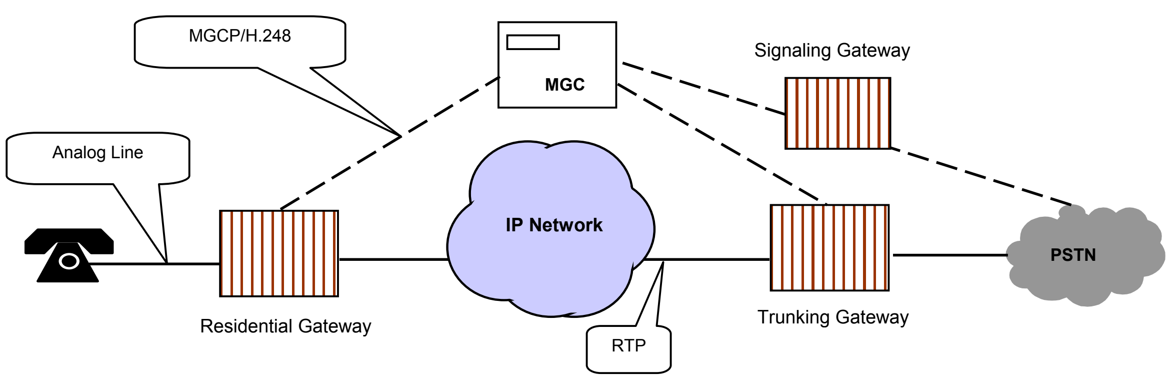
The Application Servers may communicate directly with Media Servers or through the MGC to control the MS, as well as during the call to provide advanced media services like auto–attendant, IVR, conferencing, and so on.

VoIP Tandem Switching



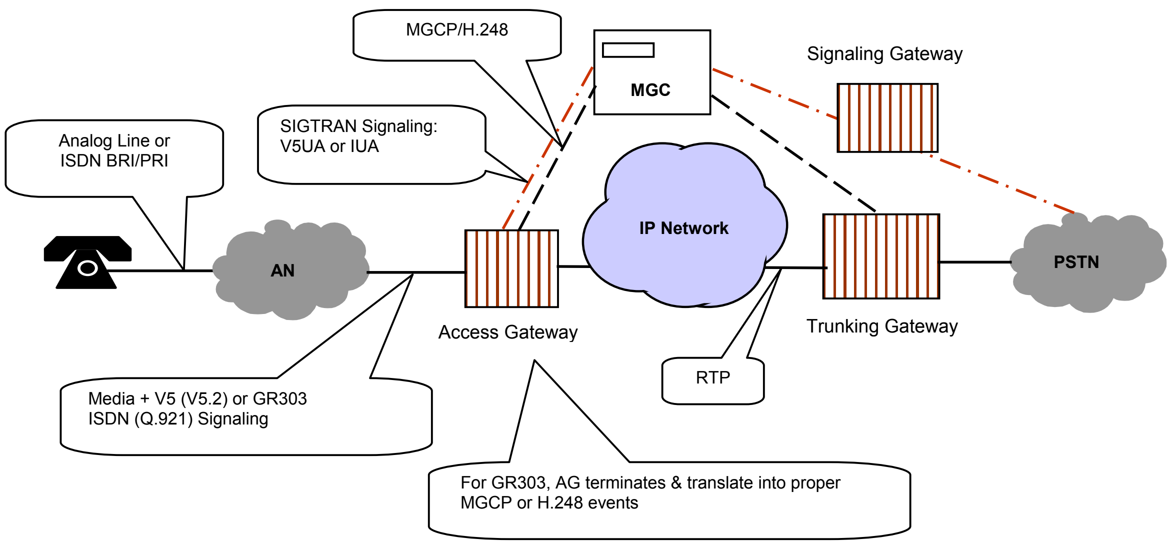
This example shows a softswitch used for tandem trunk switching, which replaces the conventional PSTN Class 4 tandem switching. The SG provides signaling transport conversion from SS7-based signaling protocols to IP SIGTRAN-based signaling protocols and transports. The TG, controlled by the MGC, provides trunking media (voice) connectivity in tandem from PSTN to IP to PSTN.

POTS Carried over IP



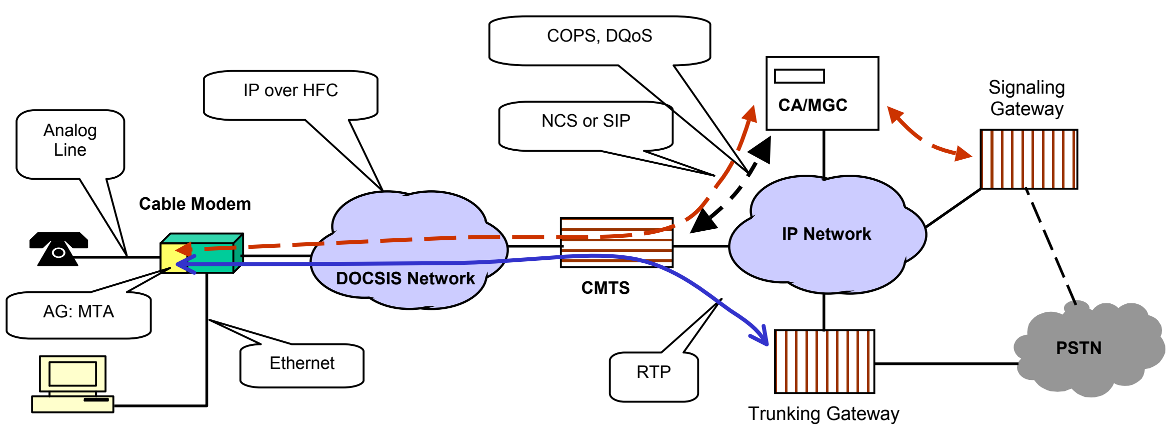
This example shows the interconnection of a POTS phone to the PSTN via the IP network. The POTS phone connects to the Residential Gateway (a type of Access Gateway). The RG performs the subscriber loop signaling and passes the signaling to the MGC using MGCP or the MEGACO protocol. The MGC in turn carries out signaling with the PSTN with the help of a Signaling Gateway. The RG digitizes and packetizes the analog voice and sends the digitized voice in RTP packets to the PSTN via the Trunking Gateway.

Access Network (V5/ISDN) over IP



This example shows a V5 or GR303 and an ISDN-based access network. The Access Gateway performs V5 or GR303 and ISDN signaling with the access network. The Access Gateway terminates the physical connection carrying the V5 or ISDN signaling and transports it over IP to the MGC using SIGTRAN (V5UA or IUA). For GR303, the AG terminates the signaling and translates it into proper events of MGCP or MEGACO for transport to the MGC. The Access Gateway packetizes and possibly transcodes the voice stream from the access network and sends the voice packets to the Trunking Gateway using RTP packets. The TG converts the packetized voice to circuit voice and transmits it to the PSTN over the physical circuit-switched trunks.

Cable Network (e.g. PacketCable™) over IP

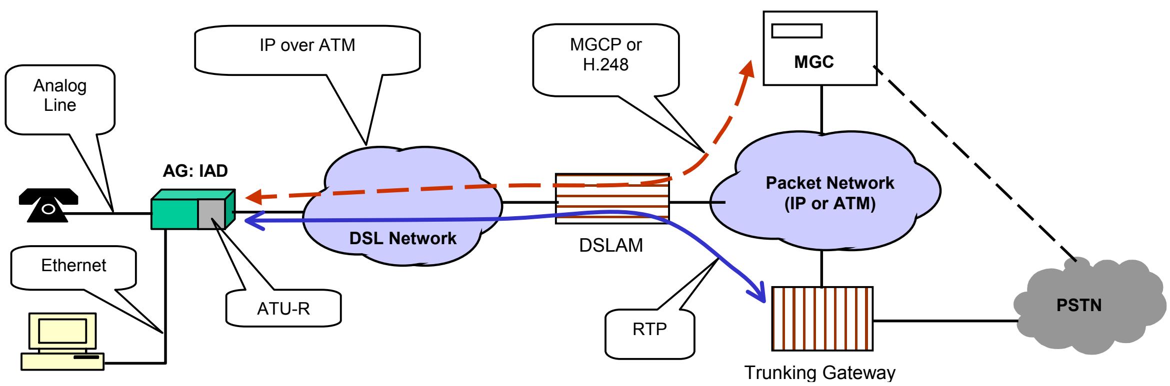


This example shows the implementation of a VoIP network using a cable access network. The cable modem at the customer premises has an embedded Multimedia Terminal Adaptor (MTA) implemented in a device called an Access Gateway (or Residential Gateway), which connects to the POTS phones and any Ethernet-based devices. An MTA can be standalone or embedded into the cable modem; a standalone MTA interfaces to the cable modem via Ethernet. The MTA terminates the subscriber loop signaling to and from the POTS phone and communicates the signaling over IP (NCS or SIP) via the CMTS to the MGC. NCS (network control signaling) is a modified form of MGCP. The MGC performs signaling with the PSTN using the Signaling Gateway. The MTA also terminates analog voice from the POTS phone, digitizes and packetizes the voice and carries it over IP via the CM/CMTS cable network to the Trunking Gateway in RTP packets. The MGC controls the TG with TGCP (a modified form of MGCP).

To be fully PacketCable compliant, the MGC-F would also have to communicate with the CMTS using signaling such as COPS.

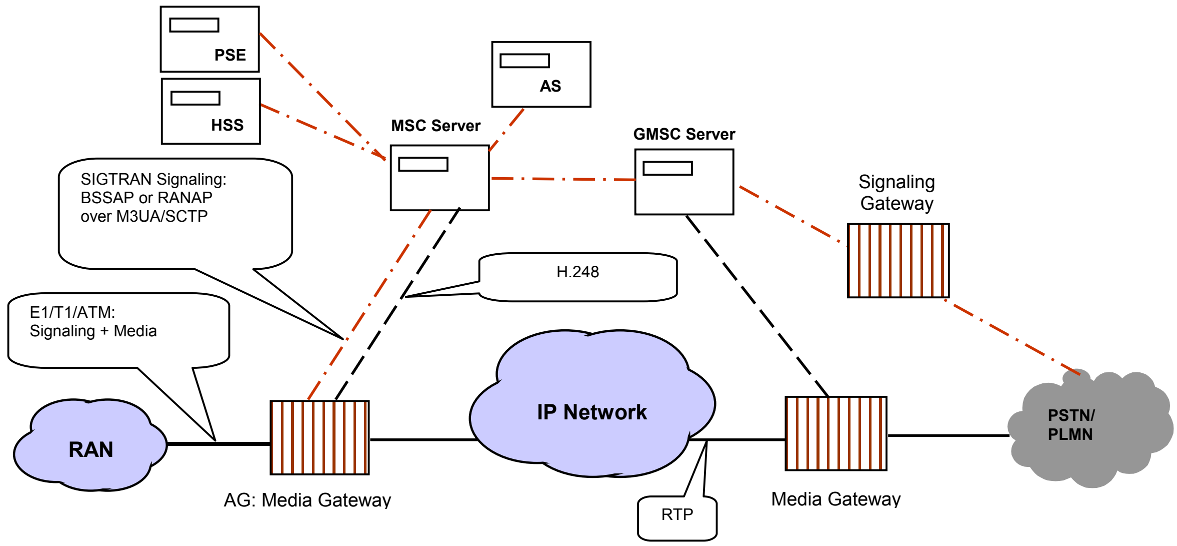
To ensure QoS when deploying VoIP over cable, the MGC communicates with the CMTS using the Dynamic QoS (DQoS) and COPS protocols.

VoDSL and IAD over IP



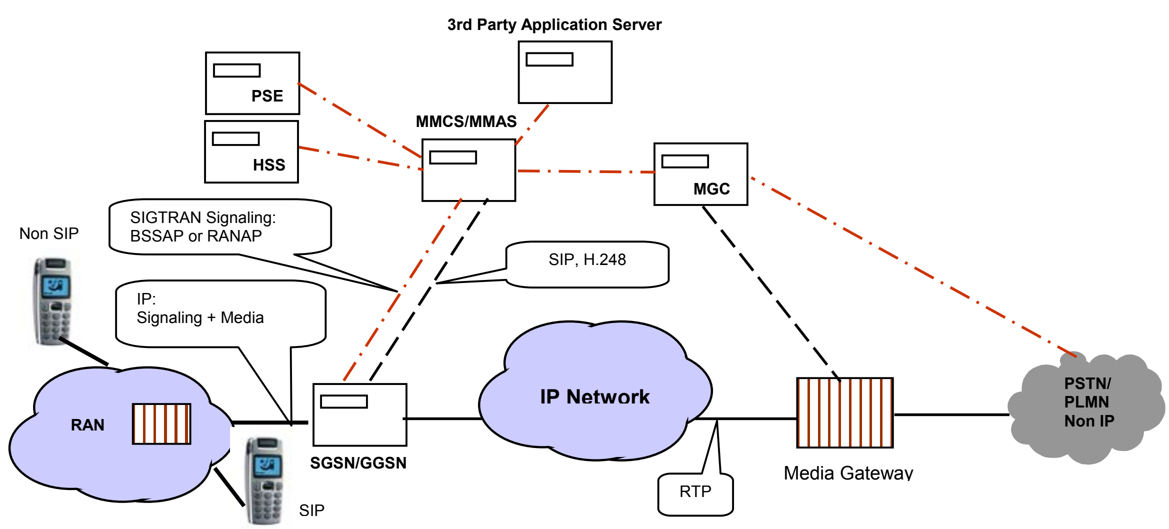
This example shows the implementation of a VoIP network using a DSL access network. The Integrated Access Device (IAD) at the customer premises (also called Access Gateway or Residential Gateway, or asymmetric subscriber line termination unit) connects to the POTS phone and any Ethernet-based devices. The IAD provides the subscriber loop signaling and communicates the signaling over IP (MGCP or MEGACO) via the DSLAM to the MGC. The MGC carries out signaling with the PSTN using the Signaling Gateway. The IAD also digitizes and packetizes the voice and carries it over IP via the DSLAM to the Trunking Gateway in RTP packets.

Wireless (3GPP R99 Special Case NGN)



This example shows how the wireless GSM/3G packet network connects to the PSTN or PLMN via the VoIP network. The Access Gateway terminates the signaling (BSSAP in GSM or RANAP in 3G) from the radio access network (RAN) over E1/T1/ATM interfaces. It transports the signaling messages over IP to the MSC Server using SIGTRAN. The MSC and GMSC servers carry out the same functionality as an MGC, including signaling with the PSTN via the Signaling Gateway, or to the PLMN via the GMSC server. The media from the RAN is terminated on the AG, transcoded and transported to the TG as RTP packets.

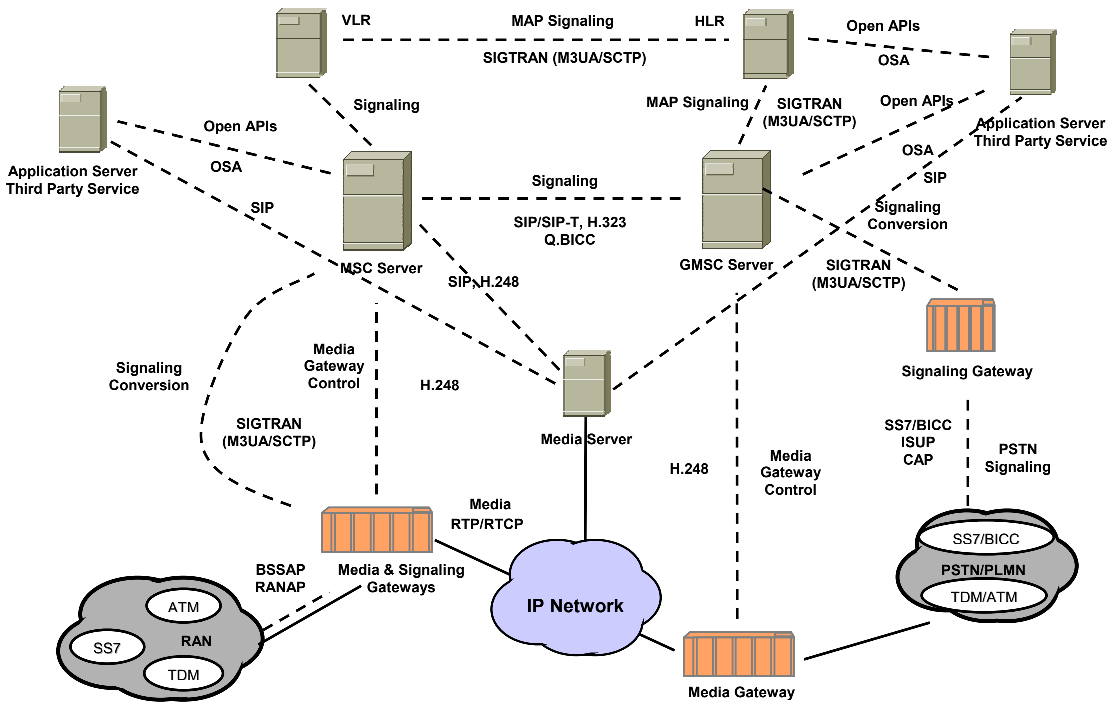
Wireless (3GPP R2000 General Case all IP)



This example shows how the wireless 3G packet network connects to the PSTN or PLMN via the VoIP network. The SGSN/GGSN passes the signaling (BSSAP in GPRS or RANAP in 3G) from the radio access network (RAN) over IP to the Multimedia Server Call Server/Application Server

(MMCS/MMAS), which provides the same functionality as an MSC Server. The MGC carries out signaling with the PSTN or legacy PLMN via the Signaling Gateway. The media from the RAN is passed through the SGSN/GGSN to the MG as RTP packets.

WCDMA Mobile Network



This example shows a total WCDMA network architecture in the circuit-switched domain, with reference to 3GPP, where an IP-based core network is used. This situation is similar to the Wireless R99 example above, but includes more complete protocols. In addition, it shows how a Media Server, controlled by the MSC Server, can provide simple announcements and under the control of an Application Server that delivers value-added services, such as voice messaging, push-to-talk and conferencing.

Appendix: Acronyms and References

This appendix contains a complete list of the acronyms used throughout the IPCC Reference Architecture. Applicable organizations and/or standards are shown parenthetically.

3G	Third Generation
3GPP	3G Partnership Project (UMTS)
AAA	Authentication, Authorization and Accounting (IETF)
A-F	(Call) Accounting Function (IPCC)
ADSL	Asymmetric Digital Subscriber Line
AG	Access Gateway
AGS-F	Access Gateway Signaling Function (IPCC)
AIN	Advanced Intelligent Network
AN	Access Network
API	Application Programming Interface
AS	Application Server
AS-F	Application Server Function (IPCC)
ATM	Asynchronous Transfer Mode
ATU-R	ADSL Terminal Unit-Remote
AuC	Authentication Center (GSM)
BG	Border Gateway
BICC	Bearer Independent Call Control (ITU Q.1901)
BSSAP	Base Station Subsystem Application Part (GSM, 3GPP)
CA	Call Agent
CA-F	Call Agent Function (IPCC)
CAMEL	Customized Applications for Mobile Network Enhanced Logic (GSM)
CAP	CAMEL Application Part (GSM, 3GPP)
CLI	Common Language Infrastructure
CM	Cable Modem
CMTS	Cable Modem Termination System (DOCSIS, PacketCable)
COPS	Common Open Policy Protocol (IEFT RFC 2748)
CPL	Call Processing Language
DiffServ	Differentiated Services
DOCSIS	Data Over Cable System Interface Specification
DQoS	Dynamic Quality of Service
DSL	Digital Subscriber Line
DSLAM	DSL Access Multiplexer
DTMF	Dual Tone/Multiple Frequency
ENUM	E.164 Numbering (IETF RFC 2916)
FS	Feature Server
GGSN	Gateway GPRS System Node (GPRS, 3GPP)
GK	Gatekeeper
GMSC	Gateway Mobile Services Switching Center (GSM, 3GPP)
GPRS	General Packet Radio Service
GSM	Global System for Mobility
HFC	Hybrid Fiber/Cable

HLR	Home Location Register (GSM, 3GPP)
HSS	Home Subscriber System (3GPP)
HTTP	Hyper Text Transport Protocol (IETF)
IAD	Integrated Access Device
IETF	Internet Engineering Task Force
IN	Intelligent Network
INAP	Intelligent Network Application Protocol
ISDN	Integrated Services Digital Network
ISUP	Integrated Services Digital Network User Part (SS7)
ITU	International Telecommunications Union
IUA	ISDN User Adaptation
IVR	Interactive Voice Response
IW-F	Inter-working Function (IPCC)
JAIN	Java Application Interface Network
LDAP	Lightweight Directory Access Protocol (IEFT)
M3UA	MTP3 User Adaptation (IETF SIGTRAN)
MAP	Mobile Application Part (GSM, 3GPP)
MEGACO	MEdia GAteway COntrol (IETF RFC 3015 or ITU H.248)
MG	Media Gateway
MGC	Media Gateway Controller
MGC-F	Media Gateway Controller Function (IPCC)
MGCP	Media Gateway Control Protocol (IETF RFC 2705)
MG-F	Media Gateway Function (IPCC)
MMAS	Mobile Multimedia Application Server (3GPP)
MMCS	Mobile Multimedia Call Server (3GPP)
MPLS	Multi-Protocol Label Switching
MS	Media Server
MSC	Mobile Services Switching Center (GSM, 3GPP)
MS-F	Media Server Function (IPCC)
MTA	Multimedia Terminal Adaptor (PacketCable)
NCS	Network Call/Control Signaling (PacketCable)
NGN	Next Generation Network
OEM	Original Equipment Manufacturer
OSA	Open Service Access (3GPP)
OSS	Operational Support System
PBX	Private Branch eXchange
PLMN	Public Land Mobile Network (3GPP, UMTS)
POTS	Plain Old Telephone Service
PSE	Personal Service Environment (3GPP)
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RAN	Radio Access Network
RANAP	Radio Access Network Application Part (3GPP)
R-F	(Call) Routing Function (IPCC)
RFC	Request For Comment (IETF)
RG	Residential Gateway
RSVP	Resource ReSerVation Protocol (IETF)
RTCP	Real Time Transport Control Protocol (IETF)

RTP	Real Time Transport Protocol (IETF RFC 1889)
SC-F	Service Control Function (IPCC)
SCP	Service Control Point
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SG	Signaling Gateway
SG-F	Signaling Gateway Function (IPCC)
SGSN	Serving GPRS System Node (GPRS, 3GPP)
SIGTRAN	SIGnaling TRANsport (IETF M3UA, IUA, SUA, V5UA Drafts)
SIP	Session Initiation Protocol (IETF)
SIP-T	SIP For Telephony (IETF Draft)
SNMP	Simple Network Management Protocol
SPS-F	SIP Proxy Server Function (IPCC)
SS7	Signaling System 7
SUA	SCCP User Adaptation (IETF SIGTRAN)
TCAP	Transaction Capability Application Part (SS7)
TDM	Time Division Multiplexing
TG	Trunk or Trunking Gateway
TGCP	Trunking Gateway Control Protocol
TLA	Three-Letter Acronym
TRIP	Telephony Routing over IP (IETF RFC 2871)
TUA	TCAP User Adaptation (IETF SIGTRAN)
UMTS	Universal Mobile Telecommunications System
V5UA	V5 User Adaptation (IETF SIGTRAN)
VAD	Voice Activity Detection
VLR	Visitor Location Register (GSM, 3GPP)
VoDSL	Voice over DSL
VoIP	Voice over IP
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access
XML	Extensible Markup Language

# # #

The International Packet Communications Consortium (IPCC) is the premiere forum for the worldwide advancement of the next generation networks through products, services, applications, and solutions utilizing packet-based voice, data and video communications technologies available today via any transport medium including but not limited to copper, broadband and fiber optics.

The IPCC establishes a common terminology for the softswitch-based architecture, and it promotes interoperability, conducts research, and liaises with governmental and industry organizations to address industry issues that service providers and vendors face. By providing a variety of educational seminars and by fostering the Open Network and Standard Interfaces, the Consortium accelerates the advancement and usage of softswitch-based networks.

The IPCC membership includes wireline and wireless service providers and carriers, governmental agencies, standards bodies, and equipment and software vendors representing all network elements involved in the softswitch-based and next generation network.