INTERTALK TECHNOLOGY

WHITEPAPER

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1. Architecture

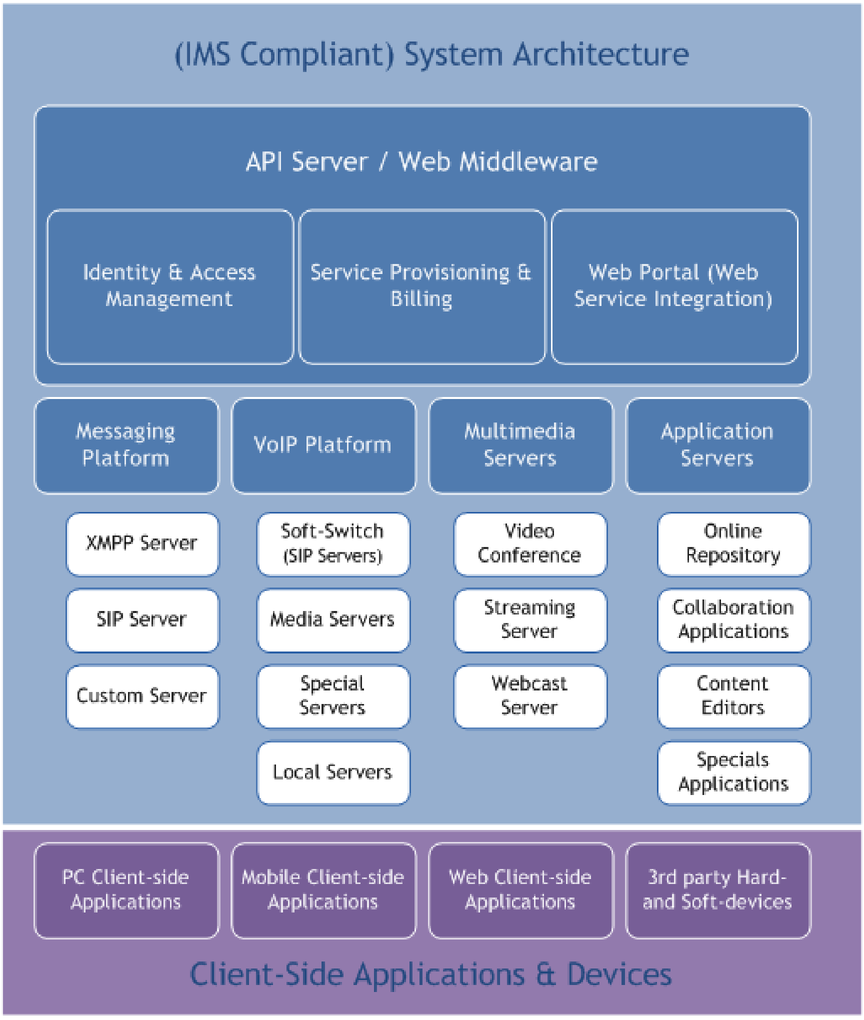
1.1. IMS & Web Service Compliance

Intertalk services are based on a proprietary software platform (developed by Intertalk R&D LAB) that is carefully architected to comply with the most recent and the most advanced new standards for Telecom and Internet services, namely with **IMS** and **Web Services** standards.

IMS stands for “IP Multimedia Subsystem” and it refers to an architecture in which various, traditionally disconnected network- and media-services re-position as layers, subsystems of an integrated IP Service platform. Typically, IMS architecture is implemented by carriers with a hope to achieve fast and efficient integration of traditionally separated voice propositions (FIXED, MOBILE & VoIP) but also in order to enable deployment of “Multi-Play” IP services (on top of existing “Dual Play” or “Triple Play”, as used to describe combined Internet + VoIP + IP TV services). The key for IMS as architecture is in separating service provisioning, billing, access management, authentication and authorization from any specific service backend, which means to create an integrated provisioning platform on top of which each specific backend is implemented as an Application server and therefore a sub-system from a point of view of the overall platform.

Web services standards and protocols enable dynamic integration of IP services, even across Web domains and service providers. They are based on a whole range of rules how to create, exchange and interpret different types of well-formed XML-files. The specs of these formats, when made available, allow custom integration with any other set of data and services. It is typical for Web services that multiple GUI forms as user-readable representations of administrative objects in the System are stored and managed as XML-based structures, with specs identifying their schemas and designs, and often with behaviour descriptions embedded in those structures. This allows dynamic form creation and on-fly adding of new system features without need for reprogramming the System. In other words – XML structures are used as templates (models) for objects and actions on those objects. As an outcome, Web Service architecture enables integration of REMOTE data, applications and services, as long as certain rules are followed and proper parsers for XML classes implemented in an adequate manner.

Intertalk platform is fully compliant both with IMS and Web service standards and protocols. It contains multiple service backend as Application Servers on top of integrated user management, access management and service billing and provisioning core. It has a whole set of parsers and renderers that are managing a whole range of different XML files, enabling dynamic service integration, even with 3rd parties, as well as easy customization and configuration of any “service-package”.



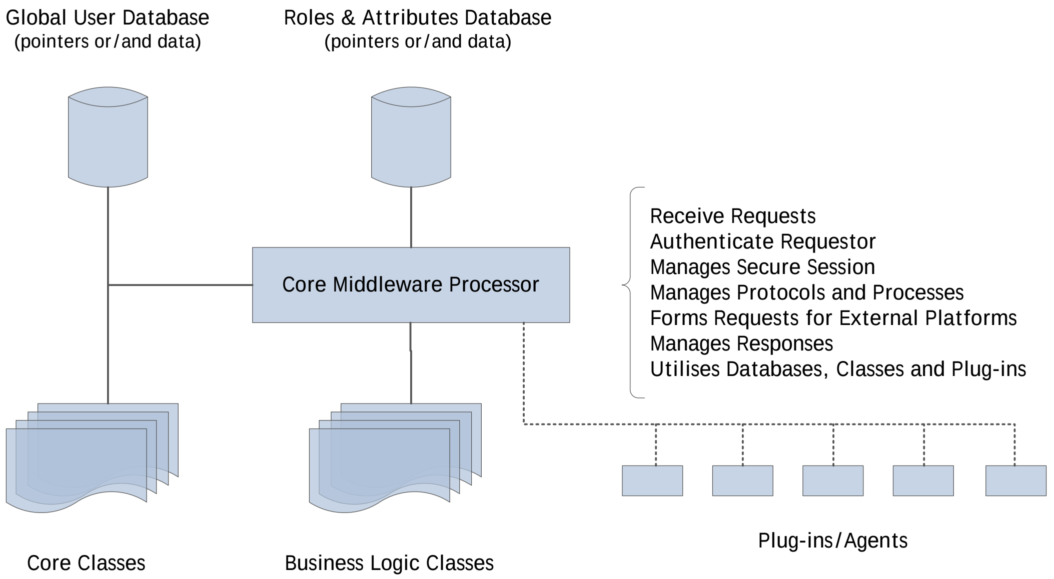
1.2. API Server as the core of Web Middleware

API Server is the key element of Intertalk Web Middleware and has an essential function for the system as a whole. It provides Application Programming Interfaces for accessing any data within the platform and, as such, it operates as a data-broker, the exclusive "agent" that delivers valuable data over a secured session to authorized parties. It is only through API Server that any other element of the platform can access data relevant for its operations. This means that API Server does authentication of the "requestor" and allows only authorized servers and services to retrieve requested data. API Server is based on (publicly available) API specifications made by Intertalk development team. These specifications create a meta-language that any (authorized) party can use to interface with Intertalk platform. This applies for all Intertalk Application Servers (for instance, for "Redirect Server" which, as we will see later, retrieves call plans from database through API Server, practically enabling call execution). But, it also applies for authorized 3rd parties – for instance, for Web Portals of Intertalk distributors and resellers who might use API specifications to configure service administration (and user self-care)internally, at their own URLs – while exchanging data real-time with Intertalk core platform by using API calls. It is obvious that such an approach enables quite some flexibility in customization as well as in delegating administration and structuring support.

1.3. Web Middleware

API Server is the key element of Intertalk Web Middleware which uses API Server to proxy specific requests and communicates to Application servers within the System. However, Web Middleware is also a meta-layer, an “umbrella” for all web-based administrative tasks. Functionalities that Web Middleware provides (together with API Server) include:

- Secured Session management
- Recognizes and generates various open formats (understands and executes XML and other open types of requests);
- Translates non-XML-requests into XML-formatted requests (using plug-ins),
- Utilizes all standard Web Service methods and protocols
- Scope of functionalities is proportional to repository of classes (which grows on daily basis) and as implemented through plug-ins to connected Back-ends/Front-ends.



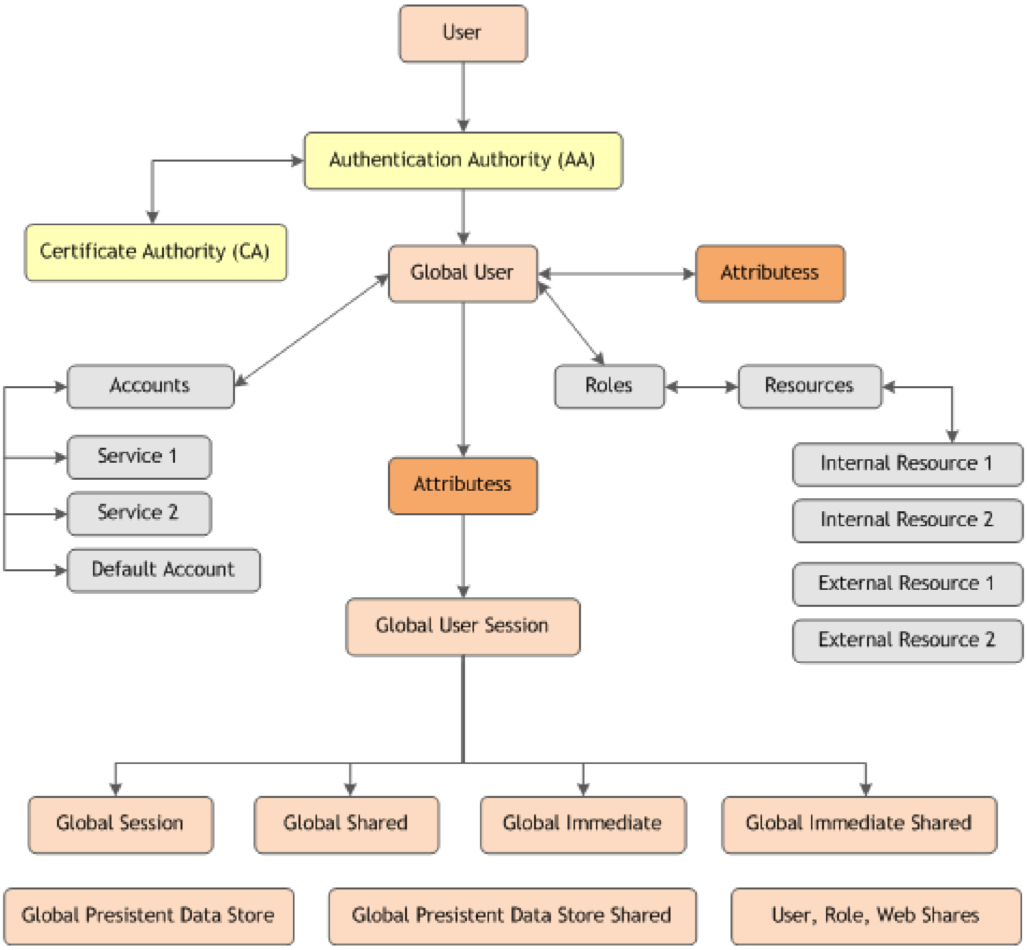
1.4. Identity and Access Management

Identity and Access Management across multiple Application Servers represents one of key Middleware functions and it is based on, so called, “**Global User**” paradigm. A Global User is a set of meta-descriptions and pointers to all User Accounts. The concept of Global User is inaugurated in order to enable preservation of independent data-sources across various Application Servers while introducing unified user web-management and identification. This makes Integration of database stores unnecessary, since those might live independently. The major shift is in introducing a new **Authentication Authority** and the “Global User” as **meta-user** of the system.

This move ensures that “private” or “global” sessions, created through any media and any application, are kept consistent and might be used for any processing logic. It is the starting point of the major architectural stone – the secure session object. Identity and Access Management, thus, relates to new, Global User category and its relation to subordinated “accounts” (data entries related to this Global

User in underlying independent data-layer systems). Identification is strict and non-distributed on Global User level and might be inherited from a particular sub-system (an Application Server).

Bellow, there is a schematic description of Global User concept as implemented for Intertalk Identity and Access Management Platform.



As the outcome, IAM solves the major issue for all integrated systems: it regulates and guarantees integrity and security of the system by assigning access and usage rights in accordance with specific **authorization** and **authentication**.

1.5. Web-based Service Administration, Provisioning and Billing

The fact that service administration, provisioning and billing are described as a part of Web Middleware confirms the basic IMS and Web Service compliant architectural approach. Intertalk service ADMIN (as popularly called both internally and by distributors and customers) is state-of-the art platform which implements so called “object model” to achieve maximal flexibility while efficiency in **implementing accounts, creating product packages, configuring service features and preferences , defining billing specifics, implementing tariff plans, assigning delegated service administration and self-care authorities to other parties etc**. The fact that this platform is totally unified and Web based, while managing configurations and accounts that are propagated in real-time to multiple Application Servers and sub-Servers, represents real proof of the feasibility of using Web service and IMS approach for IP communication services.

Intertalk ADMIN is utilizing API Server both in order to input data into appropriate databases as well as to retrieve data for various administrative purposes. Structure is fully hierarchic without any limitation as far as the number of possible levels is concerned. This means that classical tree structure (with parent-child inheritance principles) is implemented for creating hierarchies and for delegating rights (creating authorizations and sub-authorizations) and the number of nodes in the tree is fully flexible and adaptable to concrete life situations. For instance, a typical ADMIN hierarchy applicable for Intertalk VoIP services includes (from top to bottom):

- Central Intertalk administrators
- Intertalk 3rd level support at a local territory
- Distributor administrators
- Reseller's administrators
- Main corporate administrators (at customer' side)
- Individual user setting his preferences himself (a self-care mode)

But, this 6-level hierarchic structure is only one example of implementation, since there could be more or less levels dependent on services and business models.

1.6. Flexibility if generating Web Portal Interfaces (Web Service integration interfaces)

ADMIN can be distributed to several URLs as long as API calls are properly implemented and parties authorized to administer services from these IP addresses. Dependent on the IP address from which user accesses the system, different graphical elements with completely different navigation and interfaces can be implemented – enabling white-labelling and customization, while keeping functional consistency

However, this feature transcends ADMIN as such and in general applies for Web Middleware and the manner in which interfaces are generated within the system. In the Middleware, graphical interface and implementation (executing code) are separated as much as possible to allow manageable meta-layer platform:

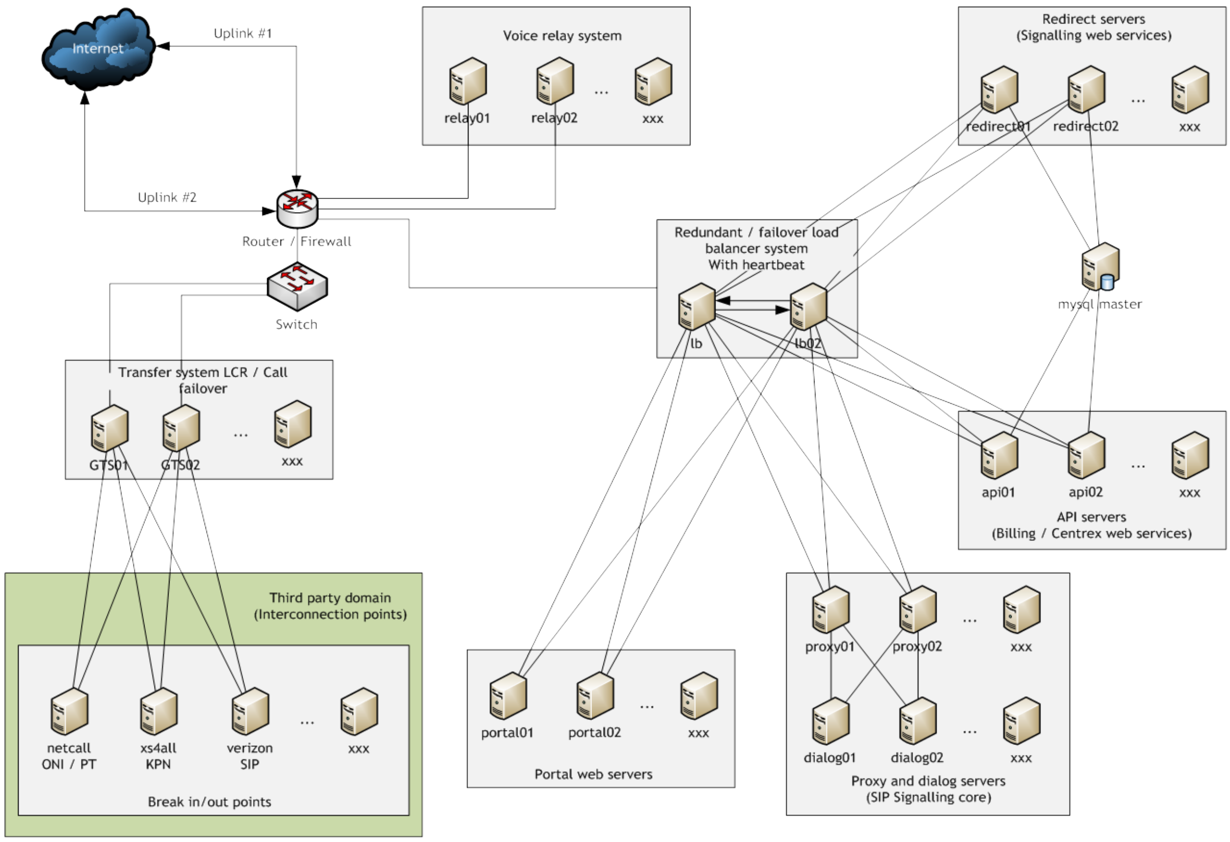
- Easy manipulation and maintenance of the GUI separately from execution code
- GUI customizations on the level of administrator (group) and individual (users) for screen, lingual, option and other purposes.
- Better exception handling and proxying the requests.
- Full encapsulation of functionalities within Java classes and separation of "common" (reusable) classes.

Middleware is listening to GUI actions, keeps the session information secure, buffers information and supplies the GUI with data needed. Morphing parameters for client-side (GUI) are all exteriorized and kept in separate XML-formed files on the web-server. Thus, client-side representation is formed fully dynamic, by taking parameters from mentioned XML files and passing them through client-side interpreter/renderer, forming the user interface on-fly.

Largest practical advantage of this approach is the fact that even client-side user-interface templates can be changed by changing only the XML "model", in addition to its dynamic forming (without changing the system). This finds its implementation also in adapting data and services to be rendered on different within different interfaces appropriate for different devices (such as PC and Mobile devices).

2. VoIP Platform

2.1. VoIP Platform Infrastructure



2.2. SIP Libraries

InterTalk VoIP platform is SIP compliant. SIP means Session Initiation Protocol and it became the new global standard for IP-based communication. InterTalk VoIP platform is using three different SIP Libraries, two of which are proprietary and one OpenSource. SIP Libraries

define how to format messages that are compliant with SIP RFC 3261 and other connected protocols. Two proprietary libraries developed in Intertalk R&D LAB, are written in JAVA and in C++. The reason for developing proprietary libraries is in the fact that existing Open-Source libraries do not facilitate high-end scalability and performance needs.

- JAVA is chosen because of its robustness and stability as well as because of its flexibility in configuring complex services and features. It is implemented for developing core Servers that are parts of the core Soft-Switch.

- C++ code path is chosen because of its speed of execution and it is used for performance-sensitive applications such as local network appliance, client-side applications on PC and Mobile, as well as for specific high-level processing servers. In other words, whenever applications run on lover-grade hardware with limited processing capacity or when processing demands on servers are extraordinary high.
- However, open-source SIP libraries, such as ASTERISK, can be safely used and are used by InterTalk, for services that are less performance-critical, such as media services. The advantage of using ASTERISK for Media Server is in the fact that there are various features and services already developed and available, which can be either simply re-used or easily re-configured to work in sync with the rest of the system. Another reason for using ASTERISK is in enabling easy interfacing and integration with 3rd party (IP PBX or client-side) solutions that are often developed in ASTERISK.

By using in parallel these three SIP libraries (JAVA, C++ & ASTERISK) with totally different advantages and by establishing easy and effortless inter-library communication, InterTalk achieves the best of all three approaches and avoids issues specific for each of these libraries (JAVA demands a lot of processing; C++ doesn't allow speedy development; ASTERISK is not robust and scalable enough).

2.3. Soft-Switch

Soft-Switch is the core of Intertalk VoIP Platform and it is designed in such a way to enable high-level vertical (load) and horizontal (new features) scalability. In order to achieve this, a network of specialized SIP Servers was developed separating whole process into individual tasks, such as: (1) authorization; (2) session-management; (3) call-plan implementation; (4) CDr generating; (5) RTP streams management; (6) transfer management; (7) interconnect management; (8) Logging and monitoring etc. Such “specialization” enables servers to process very fast their exclusive tasks getting free for processing a next request while other servers are busy with further facilitating the previous session. This segmentation and specialization essentially contributes to scalability as well as to control enabling really highest carrier-grade performance level, not so common for VoIP platforms.

Soft-Switch consists of the following servers:

2.3.1. SIP PROXY

Is the main communication point for all connected parties (clients and servers). As such, it handles, first of all, authorization of all parties, and then – if authorization was successful – distribution of calls and messages to one of available Dialog Servers. There are also several secondary tasks performed by this server, such as distribute of status-messages, and even textual messages if requested by some of the parties.

2.3.2. Dialog Server

Takes over call-related signalling once parties are authorized by SIP PROXY. This means that Dialog Server is responsible for managing concrete communication sessions. This applies not only to 1-to-1 communication but also to complex call plans that involves multiple parties (hunting, forking etc). As a secondary task, Dialog Server serves for distributing session-status messages (if requested by a certain party – for example, an operator console).

2.3.3. Redirect server

Is an "agent" that connects PROXY & Dialog servers with the central database. It delivers to Proxy everything that it needs from the database: authorization data, call plans etc. However, Redirect Server doesn't communicate directly with database, but it does it with help of **API Server** which allows (XML-formatted) remote procedure calls. Redirect server has scripting capabilities allowing to define conditional execution of requests in line with certain requirements (for instance to forbid certain execution under some specified circumstances).

2.3.4. Database

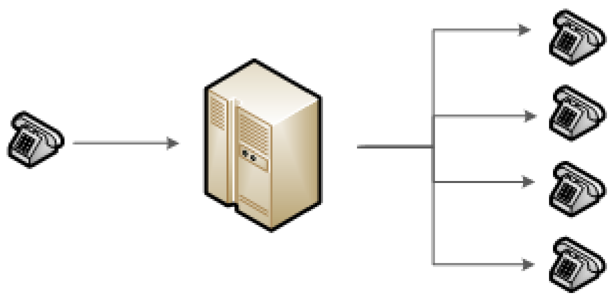
Is consisting of clustered MySQL database and it is designed as an object database with real-time backup/replication. The fact that it is an object database contributes to full flexibility in defining service packages and hierarchies, while the fact that there are (cashed, read-only)replications still enables fast real-time execution as if a procedural database were implemented.

2.3.5. Notification Server

Is responsible for notifying participants with events and statuses. Events include: keep alive (NAT session), dialog status (NAT), presence (registration status), system text messages or any custom message.

It is important to understand the logic behind such a setup of PROXY/DIALOG/REDIRECT/API/NOTIFICATION Servers, in order to fully appreciate main competitive advantages that InterTalk achieves on the VoIP Platform level.

- Intertalk SIP PROXY is a STATEFULL PROXY (in contrast to "STATELESS" PROXY, as provided by many other VoIP providers). This means that PROXY and DIALOG Servers do not only transport messages between client agents, but control and monitor states of all transactions throughout a session. It enables them to relate multiple response transactions to initial request (for instance, "INVITE" message), effectively achieving by its very architecture functionalities such as: "**forking**", "**hunting**" and other group features. It also enables implementation of most complex (and changeable/conditional) call-plans, set for each of these client-agents predicted to participate in communication. Call-plans simply define how many instances to open as a response, and PROXY does it effortlessly. However, each of these instances might have its own complex call-plan, which defines possible second layer of the transition.



- Second important architectural feature is that SIP PROXY & DIALOG are based of **PLUG-IN PRINCIPLE**. This means that the core of SIP PROXY & DIALOG is relatively simple and keeps stable and unchanged, while transactional "complications" and "specifics" (which are mostly related to call-plans) are configured as scriptable plug-ins (in php or CPL). In other words, there is a low-level **SIP API** implemented on PROXY and DIALOG, which defines how core functions can be used by internal or external scripts. Practically, this means that even the most complex call plans and conditions can be configured and the system changed by simple scripting, without complicating and damaging PROXY & DIALOG performances. This is the reason why InterTalk could offer whole range of complex call plan settings, such as: **conditional routing** (dependent on address book or any other set of preferences per caller ID), **filtering** as well as **white & black listing**, **prefix-dialling** (essential for Call Centres that facilitate various customers from the same phone devices), etc. Such and many other complex and conditional call plans can be easily add to the platform without changing core SIP PROXY and DIALOG servers.
- As previous examples show, the key architectural challenge in developing a successful and efficient VoIP platform consists in enabling **complexity and flexibility** in functionalities and processes, while keeping core of the system intact and smoothly operating. One needs to separate "**the core" from "periphery" and place specific "business logic" outside of the core**, managed either by special servers or special plug-ins. This principle can be demonstrated at work also in case of the NOTIFICATION Server, which essentially releases SIP PROXY from the need to maintain status messages (and other notifications to which certain client-agents can subscribe). For instance, in order to know the availability status of client-side devices, SIP PROXY should receive from devices KEEP ALIVE messages by means of which it determines the registration status of the devices. This however, creates tremendous load on SIP PROXY due to frequency of messaging and large number of registered devices. Taking over to itself KEEP ALIVE maintenance, NOTIFICATION Server releases the load on PROXY, contributing to the overall speed and volume of the processing. But, besides this – once NOTIFICATION SERVER was developed and configured in this specific manner – even more complex messages (including plain text messages) could be distributed to client devices, enabling complex functionalities, needed for **Operator console, BLF or SIP Messenger**.

2.3.6. Relay Servers

Manages RTP (voice and video) streams and resolve possible NAT (Network Address Translation) issues. There are two types of Relay servers: "VoIP" and "PSTN" Relays, whereby VoIP Relays are managing internal VoIP calls, while PSTN relays are used whenever at least one of the parties is external to Intertalk VoIP network. Secondary tasks of Relay Servers include quality control (monitoring and analyzing incoming and outgoing RTOP packets), pre-paid billing monitoring, fail-over etc.

2.3.7. Transfer Server

Manages two types of transfers: transfer of signals from InterTalk to 3rd party networks and transfer of signals from one to another device (number). With the first function, Transfer Server manages interconnection between InterTalk platform and external providers (mostly PSTN interconnect). It "normalizes" communication by correcting possible incompatibilities between

Intertalk messaging and messaging of 3rd party providers. With the second function, Transfer server manages blind and attended transfer call features for all calls (both VoIP and PSTN).

2.3.8. Gateways

Handling interconnect sessions with an external PSTN carrier, either as a part of a Transfer server (IP Gateway) or as stand-alone PSTN interconnect gateway (SS7, ISDN etc).

This group of servers (RELAYS, TRANSFER and GATEWAYS) represent points at which InterTalk monitors (RTP) streams, i.e. the real conversation between participants. Therefore, it is here that all stream-management or stream-monitoring tools and applications are positioned, enabling InterTalk to perform:

- Quality of Service check (even disconnect and notify users in case that quality of streams drop over a certain limit),
- Record calls (as well as Legal Intercept)
- Pre-paid monitoring (checking real-time status of prepaid accounts and disconnect calls in the moment when prepaid amount is completely spent)
- Conditional and Least-cost routing: execution of calls through chosen Gateways and interconnect partners on the basis of any preferred parameter (quality level; cost-level; fail-over activation etc).

2.3.9. Logger Server

Used for low-level SIP signalling logging.

2.3.10. Monitor Server

Hardware-level monitoring of all servers including bandwidth, processing level, disk-space, memory usage etc It also handles notification to system administrators in case of critical (miss)performances.

2.3.11. Balancer Server

Entry point into the system used to balance load to all of servers within the network. Enables hot-swop and redundancy.

2.3.12. Batch jobs

Are backend processing that facilitate various tasks: logging, invoice generation and other billing processing; price-list maintenance; CDRs backup; registration monitoring for specific clients etc

These servers and services (LOGER, MONITOR, BALANCER and BATCH JOBS) are important for enabling efficient operation management, system-data gathering and further processing (historic and real-time) as well as 3rd level support.

2.4. Media Servers

Media Servers are handling signalling and often streams for specific calls that involve either multiple participants (human or machine) or demand a specific, fully automated or semi-automated action.

Intertalk VoIP Platform includes the following services as a part of Media Server(s):

2.4.1. IVR

Enables users to set automated Interactive Voice Response to incoming calls to certain numbers, as well as to link these responses to specific call plans and conditional parameters (for instance day/time etc). Users can provide voice prompts as well as set choices and associated actions by using Web ADMIN interface.

2.4.2. Voicemail

Online storage for received voice messages as well as distribution in accordance with user's preferences (for instance, as attachment to emails).

2.4.3. FAX server

Enables to send and receive fax over Internet (with several options, including receiving fax as an email attachment).

2.4.4. Conference

Multiuser communication session management platform. Facilitates both traditional Telco-type conference (with conference rooms, appropriate codes and schedules) as well as Video conferences (where it manages authentication as well as voice part of the session). Additional features include: possibility to start an "ad hoc" conference by (automated) calling into the conference several participants at once, as well as Web based interface for managing running conference, with MUTE and KICK OF options.

2.4.5. Phone-Gate

Is a server that operates as an "access point" into Intertalk Soft-Switch for various known or unknown external devices. It is always related to specific phone numbers and specific tasks that these phone numbers are dedicated to. For instance:

- Access to Voice mail from external devices (users can call a specific number, get authenticated by Phone gate and served with recorded voice mail messages;
- Second Dial Tone, used to dial some external numbers indirectly (by using Phone Gate) instead directly from a device. This can be used for various purposes, including least cost routing and calling on the expense of company instead of private;

2.4.6. Call-Agent

Is an "avatar" of a user, which executes, as an agent, certain calls under specific circumstances. This includes:

- "Automated Calling", as for instance, "bulk calling", which means executing multiple calls from a call schedule (set by user through a Web or a Mobile interface)
- Call-back user to any of default or specified number. This is used both as a part of Web communication platform, where call-back is one of available widgets, as well as a part of GSM mobile low-cost dialling solution.
- Call continuity, to establish session while caller on one network and switch to another network if quality of the initial network connection drops (for instance, WiFi2GSM failover). In this situation, as with Call-back and Automated Calling, Call Agent is the

real caller of the call while the user who ordered the call is just one of calling party (though primary one)

2.4.7. Recording

This server reconstructs conversation from logged RTP packages and saves it in a specific folder in case of a demand either by user (RECORDING) or by authorities (Legal Intercept).

2.4.8. Music on Hold

Enables pre-recorded music (from a customer-specific folder) to calls that are held, or parked.

One of the reasons why InterTalk used ASTERISK for media server is in possibility to configure new media services effortlessly from already existing open source solution. Besides those main servers, there are several solutions that were custom implemented for specific customers, by extended ASTERISK configurations. This includes, for instance, specific Outlook and CRM integration features (TAPI-based); specific statistics requests (Wallboard); Queue management etc. On the other hand, Media services are not so performance or scaling critical, since there can be multiplied and even custom configured for specific companies or groups of companies. That's why Media servers and services are placed into local collocations at each territory where InterTalk operates, and sometimes even on customer's premises, as compounds of a local appliance (MINISIP) delivered by InterTalk.

However, the fact that this ASTERISK-based set of servers is fully synchronized with JAVA and C++ based central or local servers within Intertalk platform enabled several functionalities which are otherwise not easy to configure for plain ASTERISK implementations. This applies to CONFERENCE, where integration with Video Conference was specifically developed by InterTalk using C++ SIP library in addition to ASTERISK-based room management. This applies to CALL-GATE and CALL-AGENT where ASTERISK only handles authentication and choice of options, while calls are executed through JAVA-based Soft-Switch.

2.5. Local VoIP Appliances

2.5.1. MINISIP

MINISIP is a client-side appliance (box) that mediates between Intertalk SoftSwitch and specific devices used on a local network. These devices can be IP phones, VoIP adapters and soft-phones, or PBXs, both analogue and digital. MINISIP can be configured to function as:

- A "business continuity" or failover solution in case of problems with fully hosted service (contingency implementation);
- An IP Gateway connecting local telephony setup, even if based on analogue PBX, to Intertalk VoIP services
- A VoIP (IP) PBX, with IP devices registered directly on MINISIP, with two PSTN gateways - over a local PSTN provider and over the central Intertalk LCR (Least cost routing)

MINISIP can contain BRI and PRI connectors to PSTN, and can include also customer-specific Media servers and services.

MINISIP uses Intertalk C++ SIP library.

2.6. Client-side VoIP Solutions

In this section, only VoIP specific client side applications developed by InterTalk will be enumerated, while main bundled client-side Suites – such as **Business Communicator** and **Corporate Communication Directory** – will be separately described.

2.6.1. Configuration Wizards

InterTalk developed wizards for remote configuring and provisioning of various tested and certified client-side devices (IP phones) such as: LYNKSYS, GRANDSTREAM, SNOM and CISCO. These wizards are available in two formats: as small client-side applications that do configurations (after downloading) on local networks, as well as Web-based wizards that can do such configurations remotely.

2.6.2. PC Soft-Phone

Various versions of Intertalk soft-phone use the same C++ SIP library as MINISIP or Transfer-server and enables implementation of different graphical interfaces (from C++ to JavaScript). InterTalk soft-phone is IP CENTREX enabled, which means that it supports call transfers (blind and attended), forwards as well as other business telephony settings. Features include also Video as well as (SIP based) messaging.

2.6.3. CMA

Call Management Application, includes soft-phone as just one of its elements, on top of which it adds:

- Dual registration with two devices (CMA and a hard-phone device) with the same number
- Manipulation with hard-devices from within CMA
- View on availability status of other users, structured in various groups
- View on call status of these users (incoming calls ringing, busy status etc)
- Possibility to transfer (blind or attended) calls to other users by simple drag-and-drop
- Possibility to park calls and notify users on the calls that are parked
- Possibility to pick up parked-calls

As such, CMA can be effectively used also as an Operating console, as well as efficient call-exchange platform, useful for specific caller-groups (sales teams/support teams etc).

2.6.4. FLASH2SIP

Is a unique client-server platform that trans-codes FLASH signals into proper SIP signalling effectively enabling Web browsers to become VoIP client side applications. This enables direct Click2Dial call execution from any element of a Web page, Web Directory, address book, Web interface of a CRM etc.

FLASH2 SIP client side is available as a widget easy to integrate as a Web service into any 3rd party Web Site or Portal.

2.6.5. Mobile Business Phone

Mobile OS specific client-side application that enables:

- Directly dialling from Mobile Address book through Internet (Wi-Fi/UMTS) or indirectly through GSM, by using Phone Gate and Mobile Gateway (benefiting from Intertalk LCR capabilities)
- Usage of a mobile phone as a corporate extension, by receiving calls initially made to corporate fixed lines, transferred (either over Internet or over GSM Gateway) to a mobile device;
- Corporate call features (transfers, hunt groups etc) which truly upgrades mobile devices into proper business phones.

On top of these main features Mobile Business Phone contains Instant Massager and it can be used as accessing devices for various services included in Corporate Communication Directory.

In this moment, Mobile Business Phone is available for SYMBIAN OS, while WIN, I-PHONE, ANDROID and BLACKBERY versions are in development.

3. Messaging Platform

3.1. Architecture

As Intertalk VoIP platform is a hybrid platform, based on 3 different SIP libraries (of which one is Open Source), so also Intertalk Messaging Platform is a hybrid platform based on three different messaging protocols and servers based on these protocols (one of which is again OpenSource). Those three messaging servers are: SIP Messaging server; XMPP (JABER) Server and a Custom Intertalk IM Server. This hybrid approach was again chosen for the same reasons as in the case of VoIP platform: to avoid limitation of each of these servers, while provide complex and comprehensive services based on the benefits provided by all of them together.

3.2. SIP Messaging

SIP as a protocol can handle not only voice but also other forms of sessions, including various types of messaging sessions. Using SIP for IM has some obvious advantages for Telecoms and VoIP companies.

- Messages can be delivered to various SIP enabled devices including PC soft-clients, Mobile phones and even Fixed IP Phones.
- SIP message delivery is the fastest possible, actually real-time, since SIP is specifically design for real-time sessions.
- And finally, message delivery doesn't necessarily create any load on the server, nor require storage space, since it is directly delivered from one to another user (using servers only to establish sessions and relay the streams).

A limitation of SIP messaging is in inadequacy for IM Portal, due to the fact that Soft-Switch is used for message transfer which should not be expanded with too many features because of performance sensitivity. Therefore, another approach is needed if one wants to manage groups, group sessions, multimedia sessions etc. In short, SIP is ideal as message transfer mechanism, but not as an IM Portal framework.

3.3. XMPP or JABBER Messaging Server

Is based on an Open Source, XMPP protocol and its main advantage is in already existing network of Jabber servers all over the world, equipped with server-side connectors for all major messengers: MSN, YAHOO, GOOGLE, AMS, ICQ. Even though InterTalk developed its own client-side connectors to some of the global messengers (as MSN), JABBER-based server-side connectors are used since they allow interconnectivity with external messengers even when Web based messenger client or Mobile client are used.

But, there are several limitations of JABBER protocol, especially related to media messaging (voice and video messages included in IM session), possibility to define "special users" (for different types of corporate services) as well as group users etc. Therefore, to support these features and types of users, it was necessary to develop also a custom Intertalk IM Server.

3.4. Intertalk proprietary IM Server

Is positioned as the central IM platform that uses XMPP (JABBER) as well as SIP servers as "plug ins" or "ad on". It is based on simple HTTP protocol, and uses XML (SOAP 11) wrappers. There are several advantages of this proprietary server including:

- Adequacy for corporate communication due to the fact that http approach enables messages to pass any firewall and PROXY even at strictly protected networks.
- Since InterTalk has also FLASH-streams delivered as HTTP posts, it enables to include media (voice & video) messages into any IM session
- The fact that it is proprietary source code enables to define and implement special features as well as special types of users, effectively facilitating several specific customers' requests.

4. Multimedia Servers

4.1. Approach

Intertalk developed a whole spectrum of multimedia solutions, utilizing existing frameworks, such as Microsoft's Direct Shaw, as well as Macromedia FLASH. This enabled implementation of both PC specific as well as WEB based voice and video applications and formats. Both of these frameworks are connected to SIP servers of Intertalk, which – in case of FLASH – demanded development of a unique FLASH2SIP real-time conversion solution , effectively enabling Web browsers to become SIP VoIP devices.

4.2. Web-based Real-time Voice and Video Communication

Is enabled by FLASH Servers and Clients.

- InterTalk uses RED 5 OpenSource FLASH Server for handling simple 1-to-1 voice and video communication
- It uses Intertalk proprietary FLASH2SIP Server for executing direct calls from the Web to IP telephones or even PSTN
- And it uses a proprietary FLASH Video Conference Server (still in development) for multiuser voice and video communication over the Web.

Client-sides for all of these forms of communication are various FLASH pages, developed in-house by Intertalk FLASH experts.

4.3. Client-application based Real-time Voice and Video Communication

Uses Intertalk SIP libraries as well as existing Soft-Switch with Conference and Media Servers connected to soft-clients, such as Soft-Phone, CMA or Business Communicator. Various video formats are supported, which enables InterTalk to establish video communication between its soft-clients and some existing IP Video Hard-phones.

Voice and Video in PC Clients utilize Microsoft's DirectShow as well as Microsoft's filters for ECHO and Noise.

4.4. Video Conference

Video Conference platform of InterTalk is a very specific client-server platform. Server side is designed as a "Video Broker" and it is positioned as an "ad on" to the basic Tele-conference platform (which is a part of InterTalk Media Servers).

- This enables InterTalk to provide video conferencing totally integrated with teleconferencing: the same rooms and codes, authentication and conference management apply, disregarding of whether the conference is voice only or voice &video.
- This means that also mixed mode conferences are possible, with several participants using video conference PC client, while couple of them accessing the conference over any telephone device.
- Another advantage of such an approach is in using for conferences all existing automated and semi-automated call management and call-agent functionalities, such as call-back, which enables top organize ad hoc conferences as a part of work process, simply by calling several users INTO the conference room.

Video Sessions are managed in a very specific way. Each client-application generates two video streams, in two different sizes: full size and thumbnail-video size. Video server delivers thumbnails of all users to each video client, while full-size video is delivered per request (on client). Therefore, each participants can view all in thumbnail mode, while choose how many full-size videos he would like to have opened in any moment. In this manner, performances can be fully adapted to bandwidth conditions at the user-side.

4.5. Streaming Platform

Custom-made Streaming Server that enables delivering multimedia streams as HTTP posts.

- This enables to deliver streams even to highly protected corporate environments (since streaming is done over PORT 80 which is always opened as long as Internet browsing is permitted).
- InterTalk can deliver also real-time streams in the same manner, effectively enabling Corporate IP TV as well as Event Broadcasting.

4.6. Webcast

is a set of "media room" templates that enable users to run online events, conferences, presentations, online interviews etc. Webcast templates can be created by combining any of the following elements:

- Webcam screen (of the user who runs the event)
- Second Webcam screen (for another user whose video is broadcasted)
- Presentation screen (PowerPoint converted to FLASH)
- Files, available to download before or during the event
- The List of participants in the session
- Group chat screen (for all participants to give comments or ask questions)

Web cast can be used in PUBLIC or PROTECTED (authentication) mode and can support any number of participants (for instance, InterTalk used Webcast for back-stage interviews with musicians during TMF Awards event and in some moments over 1000 people participated in the session.

Webcast can be equipped with a payment module, and therefore can be used as a format for online professional services.

It is fully Web based and can be incorporated as a “widget” or a Web service in 3rd party’s Portals and Applications.

5. Application Servers

5.1. Net-Drive

NetDrive is a three-structured online files/folders system linked to an online repository/warehouse. Besides enabling remote access to valuable content of individual users, NetDrive also enables SHARING files with other people as well as with applications (conference, Webcast).

Each files in NetDrive is associated with specific link, which is disclosed to the user, so that he can hyperlink files (for instance recorded voice-video annotations) within other documents.

5.2. Collaboration Applications

5.2.1. Screen-Share

Enables users to transfer the content of any chosen (opened)window to other participants in a conversation, IM session, Webcast or Video Conference.

5.2.2. White-board

Multi-layered white-board that enables to draw simple objects over a background layer (which can be any file, shared screen or presentation slide) – still in development.

5.2.3. PowerPoint convertor into FLASH

Supports creation of online presentation by converting PowerPoint files into FLASH movies. Supports all animation and transitions in PowerPoint. Creates three links for each FLASH movies: STAND-ALONE link (whoever clicks can see the content); MODERATOR LINK, whatever he clicks other participants can see and PARTICIPANT LINK, can follow presentations but cannot navigate through it.

Additionally, it is possible to associate video annotation (webcam session) with each slide, effectively creating “guided presentations” which can be used as FLASH Web pages (with any content: promotional, instructional, support etc).

5.3. Online Content Editors

Open Source text and spreadsheet editors, compatible with MS office. Keep files in native HTML format, which means that can be used for managing Web content and Web pages.

6. Main Product Suites

Intertalk technology is comprehensive and complex enabling facilitation of several business models as well as packaging of various products and services. In this very moment, InterTalk promotes 3 main product-bundles:

6.1. IP CENTREX (hosted and semi-hosted business telephony)

Which is a package of business telephony features and functionalities based on Intertalk VoIP platform. This is positioned as alternative or ad on to locally hosted PBX functionalities.

6.2. Business Communicator

Which is a client-application based bundle of PC communication functionalities, including soft-phone, voice and video conference, Net Drive, IM as well as Single Sign On.

6.3. Corporate Communication Directory

This bundles Web communication tools and functionalities, including FLASH2SIP and Webcast and is structured in the way that makes it easy to incorporate in Enterprise Portals, directories and Intranet.

Technology behind these products is already described in this White Paper while more about them from the user perspective can be found in Intertalk sales and marketing documents.