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Abstract

This document presents detailed information about terminal rendering, application and protocols used in the context of SUIT project. Rendering allows the terminal to generate the images from a stream video. The application is a GUI application which is the interface between the SUIT Terminal and the end user. Protocols play an important role, thereby, permitting SUIT Terminal to communicate with the gateway in order to receive the video stream (RTP) and sending information about the terminal capabilities.

Keyword list: SUIT Terminal, Rendering, Communication Protocols, Applications

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Rendering Report

SUIT_436

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1 Introduction

1.1 Scope

The present deliverable is part of the WP6 which deals with rendering at the terminal. Rendering also includes developing GUI C++ application for an end user, with web access and VoIP.

1.2 Objective

The objective of this deliverable is to give an overview about the rendering and terminal features. Protocols used by the terminal to communicate, inform the gateway about its capabilities in respect to bit rate, display and disk space.

In the scenario where there is no gateway, the terminal shall communicate directly to the playout.

The terminal shall have support for the following protocols:

HTTP; UDP; TCP and IP as defined in IETF standards. We should point out the SUIT terminal has only been designed for testing SUIT transportation functionalities.

2 Rendering

2.1 Introduction

For rendering, we use SDL (Simple DirectMedia Layer) library which is a cross-platform multimedia library designed to provide low level access to video, and other media or devices like audio, keyboard, joystick and mouse. SDL is multiplatform which means it can be used under Linux, Windows, Windows CE, BeOS, MacOS, Mac OS X, FreeBSD, NetBSD, OpenBSD, BSD/OS, Solaris, IRIX, and QNX operation systems.

The SUIT terminal is capable of rendering the following video formats, as described in deliverable D1.1 :

- 1280x704p 16:9 -50Hz, 25 Hz
- 640x352p 25 Hz, 12.5Hz
- 320x176p 25 Hz, 12.5Hz

SUIT Terminal is also capable of decoding AAC audio coming from the playout in broadcast services or in VoD services, using digest authentication.

3 Communication Protocols

3.1 Service Discovery & Selection (SD&S)

Discovery results in a presentation of a list with services to the end user, with sufficient information for him to select a service according to his preferences.

We can have LIVE TV or VoD services. The LIVE TV services can be divided into two types :

TS full SI – This has DVB-SI integrated in the streaming.

TS optional SI – This has not DVB-SI integrated in the streaming.

The DNS server is used to locate the entry point of the Service Discovery.

The discovery of services is composed basically of two XML files according to [TS102034], in other words, is a two step approach, first the client gets service providers and then it receives the services offering for the correspondent provider.

The terminal SUIT, during the boot, access to a web address (first XML file) that points to another web address (second XML file) where is the list of services. SD&S records are delivered in segments. Segments are identified by a payload type and a segment id. The request to get a list of all services provider records with http is:

```
http://<entry_point>/dvb/sdns/sp_discovery?id=All
```

The following example shows a segment with payload id 1 containing a service provider record for the service provider SUIT. Inside the record the client finds the link to the service offerings of SUIT.

```
<?xml version="1.0" encoding="UTF-8"?>
<dvb:ServiceDiscovery xmlns:dvb="urn:dvb:ipi:sdns:2006">
<dvb:ServiceProviderDiscovery>
  <dvb:ServiceProvider DomainName="suit.org" Version="0">
    <dvb:Name Language="eng">SUIT</dvb:Name>
    <dvb:Description Language="eng">
      The SUIT project offers Live TV ...
    </dvb:Description>
    <dvb:Offering>
      <dvb:Pull Location="http://x.x.x.x:3937/dvb/sdns">
        <dvb:PayloadId Id="2">
          <dvb:Segment ID="0" Version="0"/>
        </dvb:PayloadId>
      </dvb:Pull>
      <dvb:Push Port="3937" Address="224.0.23.14">
        <dvb:PayloadId Id="5"/>
      </dvb:Push>
    </dvb:Offering>
  </dvb:ServiceProvider>
</dvb:ServiceProviderDiscovery>
</dvb:ServiceDiscovery>
```

The service provider discovery record defines the service provider (and its existence) and contains the link to the actual service discovery records. In the next example, the locations for the offerings services are used, just to show that it is possible to mix unicast and multicast delivery. Segments with the payload type 2 (broadcast discovery) are delivered in unicast mode and segments with payload type 5 (package discovery) with multicast. For the unicast delivery all segments have to be listed in the record, because there is no other way for the client to know about a segment.

In the second step the client receives the service offerings. In the next example the client will get the BroadcastDiscoveryRecord with the http request:

http://x.x.x.x/dvb/sdns/service_discovery?id=suit.org&Payload=02&Segment=0000

The server returns a XML document that contains the broadcast discovery record with a list of services and their properties.

Example:

```
<?xml version="1.0" encoding="UTF-8"?>
<dvb:ServiceDiscovery xmlns:dvb="urn:dvb:ipi:sdns:2006"
xmlns:tva="urn:tva:metadata:2005">
  <dvb:BroadcastDiscovery DomainName="suit.org">
    <dvb:ServiceList>
      <dvb:SingleService>
        <dvb:ServiceLocation>
          <dvb:IPMulticastAddress Port="5000" Address="230.1.2.3"/>
        </dvb:ServiceLocation>
        <dvb:TextualIdentifier ServiceName="suit_tv"/>
        <dvb:DVBTriplet ServiceId="40000" TSId="9999" OrigNetId="1"/>
        <dvb:SI ServiceType="16">
          <dvb:Name Language="eng">SUIT TV</dvb:Name>
        </dvb:SI>
        <dvb:VideoAttributes>
          <tva:Coding
href="urn:dvb:ipdc:esg:cs:VideoCodecCS:9"></tva:Coding>
          <tva:Scan>progressive</tva:Scan>
          <tva:HorizontalSize>1280</tva:HorizontalSize>
          <tva:VerticalSize>704</tva:VerticalSize>
          <tva:AspectRatio>16:9</tva:AspectRatio>
          <tva:Color type="color"></tva:Color>
          <tva:FrameRate>25</tva:FrameRate>
        </dvb:VideoAttributes>
      </dvb:SingleService>
    </dvb:ServiceList>
  </dvb:BroadcastDiscovery>
</dvb:ServiceDiscovery>
```

The example signals one HD service. The service is identified by a textual identifier which is **suit_tv.suit.org**. The actual service name for display to the user is “SUIT TV”. The service type is 0x16 (hex!), means a service using an advanced codec. The exact codec type is defined in the VideoAttributes element. Here the codec is MPEG4-SVC. The alternative here is VC-1. If the Coding element is omitted the video codec is MPEG2, but in that case also the service type has to be different: 0x1. The example shows all possible video attributes in the schema, like resolution and aspect ratio.

The service location element contains the link to the media stream which is either a multicast address or a RTSP URL. The example service above is multicast.

An sequence diagram of the terminal start-up is shown in section 5.2

3.1.1 SD&S Transport Protocols

The SD&S information are transported using the protocol DVBSTB in the multicast scenario, or HTTP in the unicast scenario.

The protocols are defined in [TS102034].

3.2 Session Management Protocols

For session management protocols, the terminal uses the following:

IGMP v3 [rfc3376]– for multicast

RTSP/RTP/RTCP [rfc2326] – for unicast

A brief description of the implementation protocols according to deliverable 4.2 follows:

The service can be accessed by the terminal through IGMP [rfc2236],[rfc3376] or RTSP [rfc2326]. The Live TV is broadcasted under multicast IP, but it can also be broadcasted under unicast IP. In IGMP, the streaming is continuous and it doesn't need to be started by the terminals. The terminal can join and leave the multicast group, for that purpose the terminal shall send an IGMP JOIN and LEAVE. The attribute of the service location, present in the SD&S message, gives the terminal all the information to generate the IGMP appropriate messages.

The RTSP is used to control the VoD service, also LIVE TV, play, stop and pause. The SD&S provides to the terminal the access to a location for access to the desire service. In the case where the access to the service is by RTSP, the location should be given by an SDP file. This file has, amongst other information, an URL for an RTSP server. The RTSP messages between the terminal and RTSP server (Playout), via the gateway, uses persistent TCP which permits the RTSP server to send asynchronous messages to the terminal, even if it is firewall protected.

The video stream is encapsulated in RTP, so it can be sent by the RTSP server in both modes, unicast and multicast. In multicast scenario the control operations like pause can not be used.

To access to DVB-IP the following methods for transport management and streaming control are used:

➤ SEND :

SETUP; OPTIONS; PAUSE; PLAY; TEARDOWN

➤ RECEIVE :

REDIRECT

All this methods are described in Deliverable 4.2.

3.3 Session and Services Description

SDP [rfc4566] provides a standard representation for such information, independent of how the information is transported. SDP is purely a format for session description, it does not include a transport protocol and is intended to use different transport protocols as appropriate.

RTSP [rfc2326] services are obtained from SDP file.

Thus SDP information includes:

Session name and purpose

Time(s) the session is active

The media comprising the session

Information to receive those media (addresses, ports, formats and so on)

Session description

v= (protocol version)

o= (owner/creator and session identifier).

s= (session name)

i=* (session information)

u=* (URI of description)

e=* (email address)

p=* (phone number)
c=* (connection information - not required if included in all media)
b=* (bandwidth information)
One or more time descriptions (see below)
z=* (time zone adjustments)
k=* (encryption key)
a=* (zero or more session attribute lines)
Zero or more media descriptions (see below)

Time description

t= (time the session is active)
r=* (zero or more repeat times)

```
v=0
o=- 7885924322 1 IN IP4 10.0.1.1
s=suit_tv.sdp
i=<no author> <no copyright>
c=IN IP4 0.0.0.0
t=0 0
a=type:broadcast
a=control:rtsp://x.x.x.x:554/rtpencoder/suit_tv/suit_tv.sdp
a=range:npt=0-
a=x-qt-text-nam:SUIT TV
a=x-qt-text-inf:suit_tv
m=video 10000 RTP/AVP 96
a=rtpmap:96 H264/90000
a=fmtp:96 packetization-mode=1;profile-level-id=4D4033;sprop-parameter-sets=Z01AM5p0CwS0IAAAACAAAZR4wZU,aO4G8p==
```

3.4 Feedback Channel Protocols

The terminal use MPEG-21 DIA UED to inform the gateway aspects of its environment which are described in a XML Schema according to Deliverable 4.2. This is an important information which allows the playout to identify the capabilities of the terminal and in scenario where the disk has not sufficient saving space the playout shall reject the request from the terminal to record the video. RTCP supports QoS exchange between the gateway and the terminal and provides additional bitstream adaptation to adjust bitrate to the conditions of the local WLAN.

4 Hyperlinked Video

4.1 Description

Hyperlinked video is a video spot in which specific objects are made selectable by the user. Hyperlinked video will provide an interactive service providing PIP when the user is watching a full screen image. To render a new hyperlink video, we also use SDL library as rendering tool. The following figure shows the hyperlinked video that is provided by SUIT Terminal.



Figure 1. SUIT HyperLinked Video

5 SUIT Terminal

5.1 Introduction



Figure 2. SUIT Terminal

The figure above shows the end user terminal running. In the screen, it is possible to watch the services provider on the right side, in this example the service provider only has few services, but can obviously have many more. The terminal shows the current service on the left side, the service is rendered according to the graphical display set in the SUIT terminal. These are the steps for the terminal to render the video, after processing all steps as described in Section 3. Then SUIT Terminal receives via WiFi network the RTP stream, which is decapsulated, decoded using SVC decoder and finally rendering. SUIT Terminal is capable of playing local videos, this will be used when the end user records videos and would like to watch them afterwards. The below figure 3 shows a full screen example.

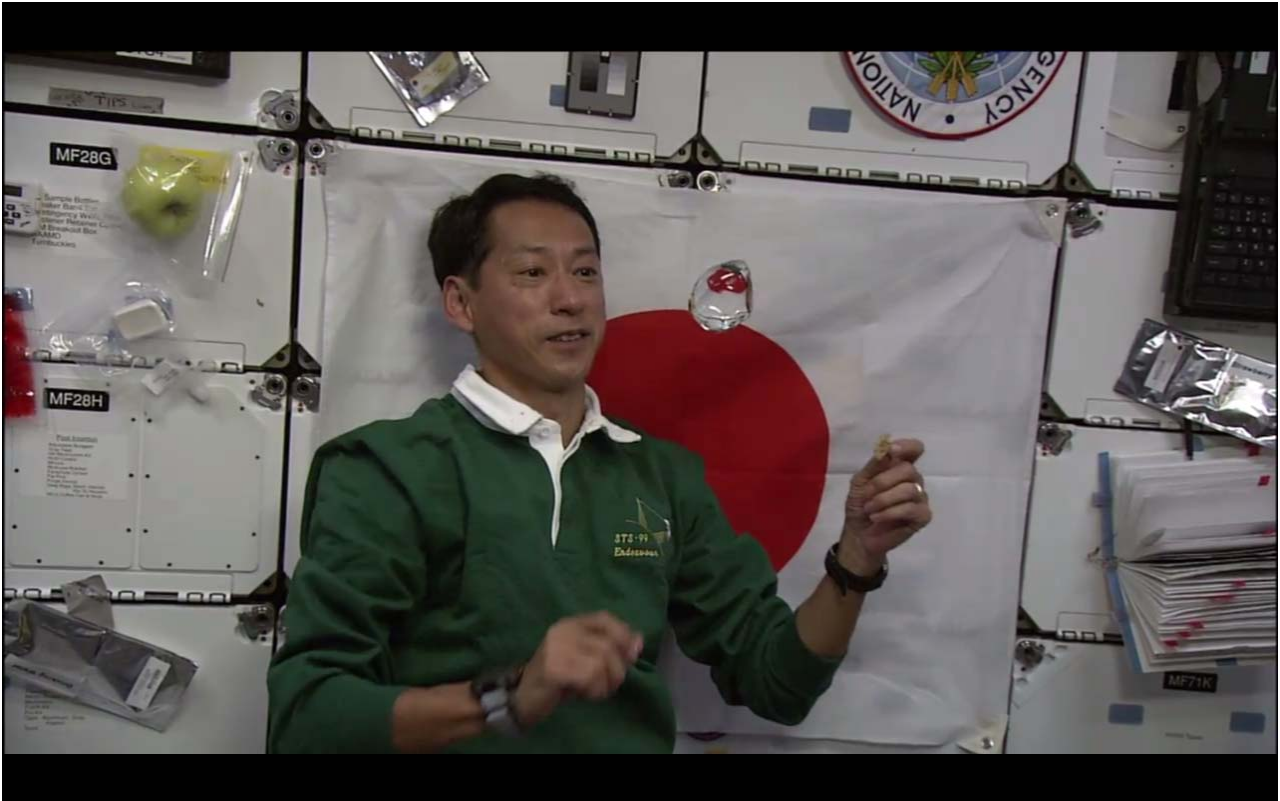


Figure 3.SUIT Terminal Full Screen

Internet Access and VoIP using the Internet Explorer and Skype have been tested. An integrated application for Internet Access and a VoIP proprietary solution are undergoing, and should be fully implemented.

5.2 Sequence diagram of the terminal start-up

As explained in chapter 3, we will show a sequence diagram of the terminal start-up. The sequences diagrams below show the steps that are taken when the terminal is executed. First the terminal finds the SD&S service, executing the SD&S service discovery in order to locate the SUIT playout services. After the terminal receives the services list of the services that are available at the playout, the end user will choose the programme that would like to see in the terminal by clicking on the provider list that appears on the right side of the terminal. Finally, depending on the provider service the terminal could join a session (IGMP) shown in figure 4 or request a new session (RTSP) as shown on figure 5.

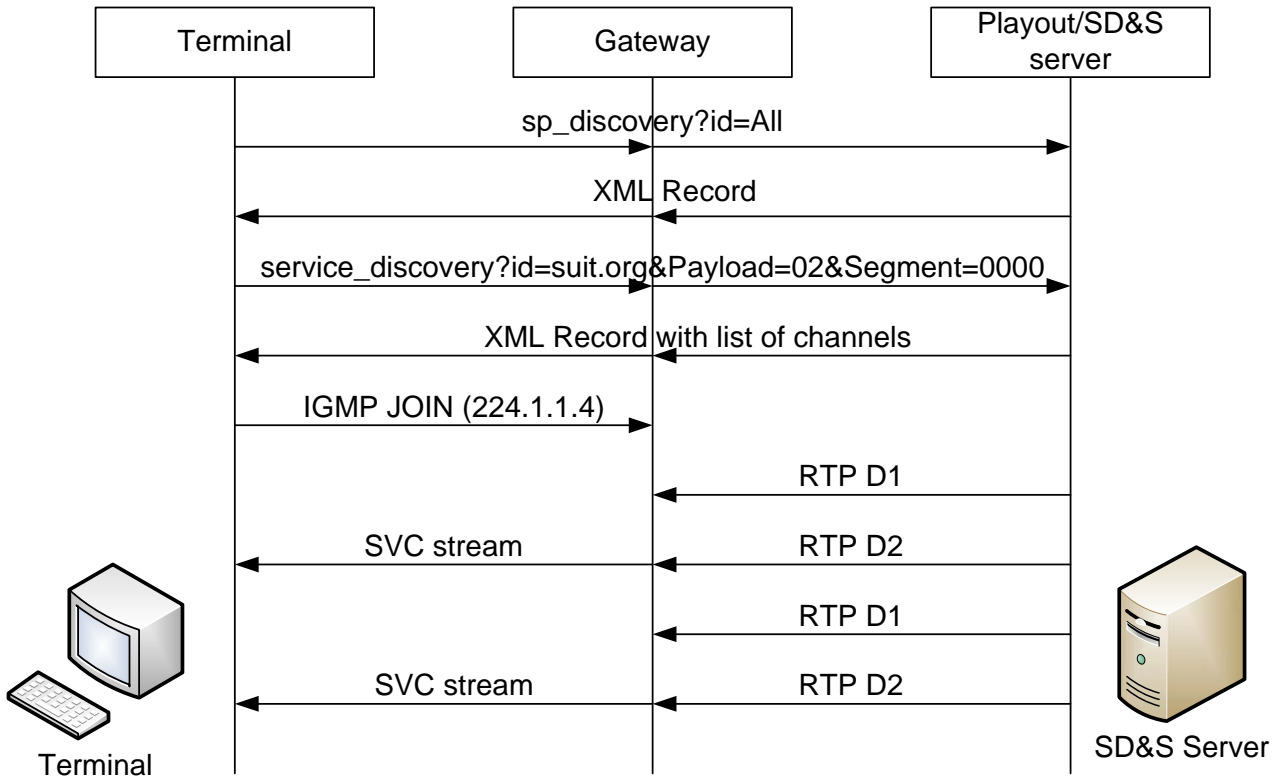


Figure 4. Sequence Diagram of Terminal Power-Up in a multicast scenario

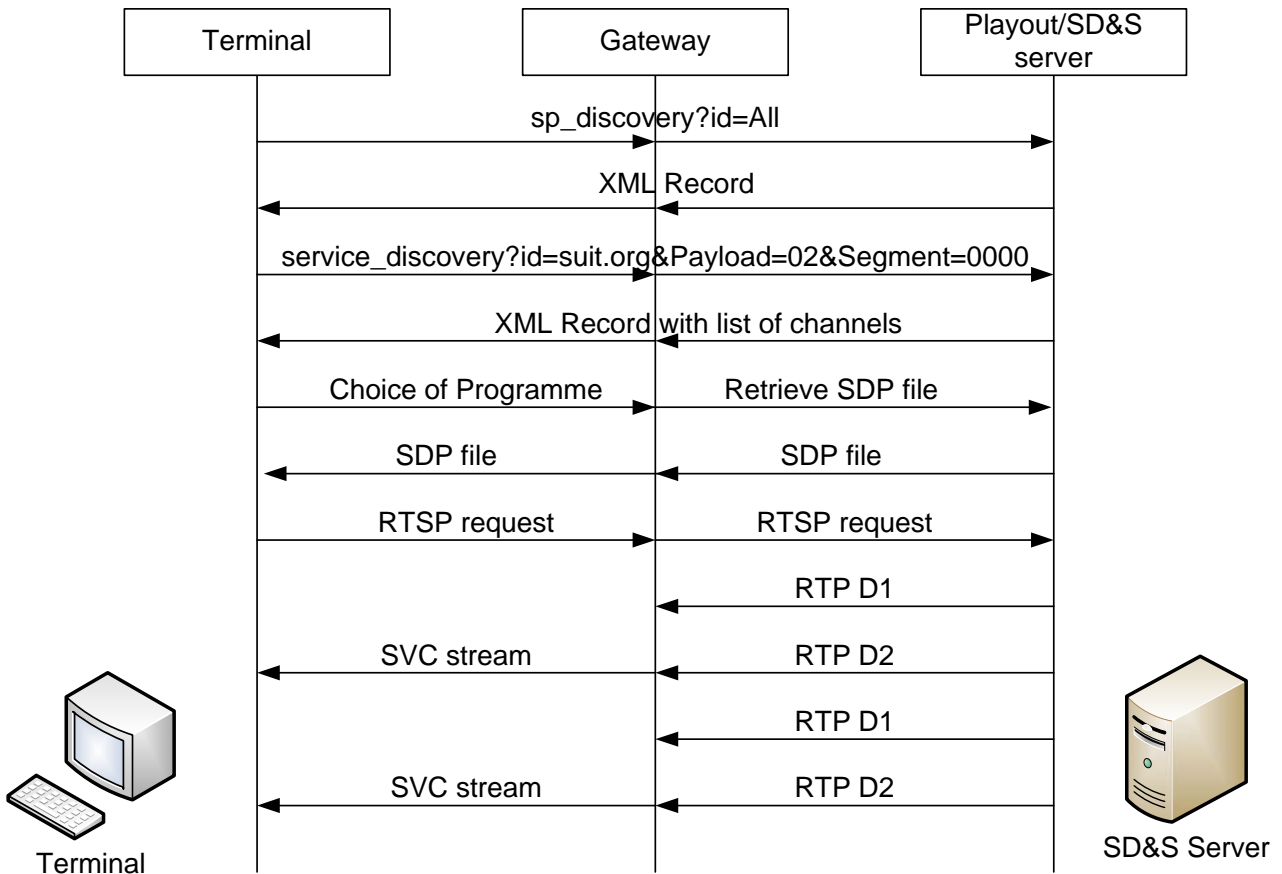


Figure 5. Sequence Diagram of Terminal Power-Up in a unicast scenario

6 Conclusions

This deliverable gives an overview about the rendering at the terminal and the protocols that the GUI application uses to communicate with the gateway.

The most complex part for SUIT Terminal is the rendering video with video formats above CIF. However, we selected a processor (dual core, T7200@2.00 GHz) which is able to decode and render HD video formats. The terminal is depicted in figure 6, below. SUIT Terminal is flexible since it runs under windows XP.

The delay, between the instant of time that the user selects the program to view and the time the image and audio play, is about 2 seconds.

Since SVC uses multiple layers, so in order to decode HD layer the decoder gets information from the other two layers (SD,CIF). This means that if there is some packet loss in the network and/or in the radio links there can be some freezes in the process of rendering image due to loss of NAL units.



Figure 6. Terminal Hardware

7 Acronyms

AAC	Advanced Audio Coding
AVC	Advanced Video Coding
CIF / QCIF	CIF (Common Intermediate Format) is used to standardise the horizontal and vertical resolutions in pixels of YCbCr sequences in video signals. QCIF means "Quarter CIF".
DI	Digital Item
DIA	Digital Item Adaptation
DID	Digital Item Declaration
DTV	Digital Television
DVB	Digital Video Broadcast
DVBSTP	Digital Video Broadcast – SD&S Transport Protocol
DVB-T	Digital Video Broadcasting - Terrestrial
ETSI	European Telecommunications Standards Institute
GUI	Graphic User Interface
HTTP	HyperText Transfer Protocol
IANA	Internet Assigned Numbers Authority
IEC	International Engineering Consortium
IETF	Internet Engineering Task Force
IGMP	Internet Group Management Protocol
ISO	International Organization for Standardization
MDC	Multiple Description Coding
MPEG	Moving Picture Experts Group
PCI	Peripheral Component Interconnect
PIP	Picture in Picture
QoS	Quality of Service
RFC	Request For Comments
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
SAP	Session Announcement Protocol
SD&S	Service Discovery and Selection
SDP	Session Description Protocol
SVC	Scalable Video Coding
UED	Usage Environment Description
VoD	Video On Demand
VoIP	Voice over Internet Protocol
W3C	World Wide Web Consortium
WLAN	Wireless Local Area Network
WS	Web Service
XML	Extensible Mark-up Language

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