VOO
Free Modem Choice
Network Termination Point Interface
VOICE Spec's

History

Version	Description
1.x	Draft for comment- First Version

Table of contents

Acronyms :

- ABNF Digit Map String representation (“Augmented Backus-Naur Form”)
- ATA Analog Terminal Adapter (FXS port with VoIP)
- DHCP Dynamic Host Configuration Protocol
- DNS Domain Name System
- DSx Dynamic Service flow Add/Change/Delete
- DTMF Dual-tone multi-frequency
- EDVA Embedded Digital Voice Adapter (PacketCable 2.0 device)
- eMTA embedded MultiMedia Terminal Adapter:
  a modem with built-in voice capability (FXS port)
- IMPU IP Multimedia Public User
- IMPI IP Multimedia Private Identity
- IP Internet Protocol
- IPv4 Internet Protocol version 4

- IPv6 Internet Protocol version 6
- MTA MultiMedia Terminal Adapter
- MWI Message Waiting Indicator
- NA(P)T Network Address (and Port) Translation
- NAPTR Name Authority Pointer
- P-CSCF Proxy Call Session Control Function
- QoS Quality of Service
- RST Residential Services Telephony (PacketCable 2.0)
- RTCP Real-time Transport Control Protocol
- RTP Real-time Transport Protocol
- SBC Session Border Controller
- SCTP Stream Control Transmission Protocol
- SDP Session Description Protocol
- SIP Session Initiation Protocol
- TCP Transmission Control Protocol
- TLS Transport Layer Security
- TTL Time To Live
- UA User Agent
- UDP User Datagram Protocol
- UE User Equipment
- URI Uniform Resource Identifier
- VSC Vertical Service Code

Part 1 : General

Introduction

Following the BIPT Decision of 26 September 2023 regarding the identification of the network termination point (NTP) for broadband services and TV services, VOO defines in this Voice NTP document the specifications that need to be fulfilled by a Private Modem for use on the footprint of the **VOO** cable network.

DISCLAIMER

The document is based on the current state of information and network specifications and is subject to change.

The specifications may change if deemed necessary and may break backward compatibility with previous versions. When a new version of the document is being published all previous documents become void, in line with any applicable delay period.

Major effort has been put into making this document as complete and accurate as possible, nonetheless VOO cannot be held liable for any direct, indirect, incidental, consequential, or special damages arising out of the use of the information.

The interface specifications do not apply under abnormal operating conditions such as:

- Operating conditions resulting from the use of services other than DOCSIS 3.1 over the dedicated data RF interface.
- Operating conditions arising from faults, maintenance, construction work, or efforts to minimize service interruptions.
- Operating conditions resulting from force majeure or third-party interference.
- Operating conditions during test signal injection governed by regulation.
- Instances of non-compliance with the relevant standards by an End-User's installation, equipment, or technical requirements for connection, as established by this interface specification or public authorities, including the specified limits for electromagnetic compatibility.

The characteristics given in this interface specification are intended to be used to derive and specify requirements for equipment such as coaxial cables and cable modems to connect them to the dedicated data RF interface.

The values in this interface specification take precedence over requirements in equipment product standards and installation standards. The given characteristics are not intended to be used as electromagnetic compatibility levels or user emission limits in the HFC networks.

CPE Supplier must provide the right information, support, and software updates to the end-customer when end-customers use their own terminal equipment (Private Modem). It is the CPE Supplier's responsibility to provide the End-user all required information to do the installation correctly, to ensure the End-user uses the most recent software updates (including security software) and to make sure that the Private Modem complies with the specifications described in this document. It is therefore required that the CPE Supplier provides the right information, support, and software updates to the customers of its devices.

VOO cannot be held liable for any problems arising out of the use of a Private Modem which does not comply with industry standards (e.g. for security) or with the present specifications. Moreover, VOO reserves the right (in line with the BIPT decision) to block any Private Modem which has a negative impact on the network.

VOO accepts no responsibility or cannot be held liable in any way for non-compliance with legal or regulatory requirements due to the use of Private Modems which are not compliant with the most up to date NTP-specifications.

VOO provides services over cable networks that are subject to technological evolution. Parts of the cable networks used for the VOO services will or may be replaced by fiber-based networks in the near and/or medium-term future. This implies that modems built for services over cable networks may no longer be usable at locations where a fiber network is deployed. As the speed and geographical scope of this evolution depends on a variety of parameters, VOO cannot provide guarantees regarding the duration during which a Private Modem based for services over cable networks can effectively be used at a given location. Suppliers of Private Modems based on these NTP specifications should warn customers of this uncertainty.

Should the end customer decide to change service provider while retaining his own Private Modem, OBE cannot guarantee that the services of the new service provider will be supported (Packet Cable VoIP, ...). It is the CPE Supplier's responsibility to inform its customers regarding the constraints applicable to its devices.

Similarly, when End-users move from one location to another while remaining a customer of OBE, OBE cannot guarantee that the services at the new location will be supported by the Private Modem. It is the CPE Supplier's responsibility to either ensure compliancy with the different services on different locations following the NTP requirements defined by the operators, either to inform its customers adequately and completely regarding the geographical and service constraints applicable to its devices. See also "Roles & Responsibilities" further in this document.

As reminder:

Operator	VOO	Orange North	Orange South

	voo	orange	orange
Voice Services	PacketCable 2.0 (built-in Voice FXS port(s) used)	Over the top ATA Cisco Box	Over the top ATA Cisco Box

Specifications update

VOO will update the present specifications whenever:

- There is a significant network change that requires an evolution of the present specifications (V-CCAP, Docsis 4.0, new overlay model, new frequency plan, others ...)
- The present specifications have been found to not be sufficiently accurate or exhaustive to meet the aim of the present specifications. Amendments to the present specifications will be issued to address such inaccuracies or lacking elements.

Convention

The capitalized terms below have the following meaning in this document :

"MUST": This word means that the item is an absolute requirement of this specification.

"SHOULD": This word means that, except valid reasons in particular circumstances to ignore this item, this item is highly recommended.

"MAY": This word means that this item is optional.

The main definitions in these specifications are :

- **"Cable Operator"**: Voo HFC network Operator (in this document).
- **"Private modem"**: Modem, other than the modem provided by VOO, chosen by the End-user to make use of the VOO services via the Cable Operator network. This will be synonym of CMP ("Cable Modem Propriétaire"), third-party modem & proprietary modem.
- **"CPE Supplier"**: supplier of a modem intended for use as Private Modem.
- **End-user :** a customer of the OBE retail services on the Cable Operator network.

Scope

The present NTP document is the implementation of the BIPT Decision of 26 September 2023 defining the NTP for broadband services related to OBE services provided over the VOO cable network.

The BIPT Decision refers to **NTP location A** according to the scheme below, which means the End-User must have right to use the modem of his/her choice to make use of the VOO services.

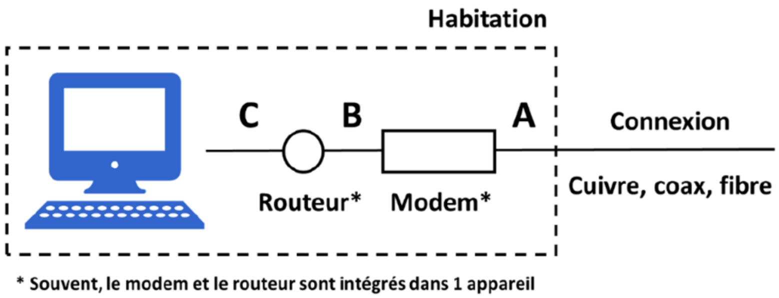


Figure 2 : Les différentes localisations possibles du NTP pour un service d'accès à l'internet.

As mentioned, the VOO Voice service is provided on the VOO network. Consequently, the specifications of the VOO underlying network must be considered.

PART 2: PacketCable 2.0 Technical Part

References

In the case of a conflict between specific requirements in this document with requirements in any of the directly or indirectly referenced documents, the specific requirements of this document are applicable.

PacketCable™ 2.0 References

Here below the last list of technical reports & specifications of packetCable 2.0:

Accounting Specification	PKT-SP-ACCT-C01-140314
Codec and Media Specification	PKT-SP-CODEC-MEDIA-C01-140314
Control Point Discovery Interface Specification	PKT-SP-CPD-C01-140314
Electronic Surveillance Delivery Function to collection…	PKT-SP-ES-DCI-C01-140314
Electronic Surveillance Intra-Network	PKT-SP-ES-INF-C01-140314
E-UE Provisioning Data Model	PKT-SP-EUE-DATA-C01-140314
E-UE Provisioning Framework	PKT-SP-EUE-PROV-C01-140314
Presence Specification	PKT-SP-PRS-C01-140314
Quality of Service	PKT-SP-QOS-C01-140314
Architecture Framework Technical Report	PKT-TR-ARCH-FRM-C01-140314
HSS Technical Report	PKT-TR-HSS-C01-140314
NAT and Firewall Traversal Technical Report	PKT-TR-NFT-C01-140314
Quality of Service Architecture Technical Report	PKT-TR-QOS-C01-070925
Security Technical Report	PKT-TR-SEC-C01-140314
SIP Signaling Technical Report	PKT-TR-SIP-C01-140314

Normative References

1. [G.168] ITU-T, G.168 : Digital network echo cancellers
2. [G.711] ITU-T, G.711 : Pulse code modulation (PCMA) of voice frequencies
3. [RFC1034] Domain Names – Concepts and Facilities

4. [RFC1035] Domain Names – Implementation and Specification
5. [RFC2131] Dynamic Host Configuration Protocol
6. [RFC2617] HTTP Authentication: Basic and Digest Access Authentication
7. [RFC2782] A DNS RR for specifying the location of services (DNS SRV)
8. [RFC2915] The Naming Authority Pointer (NAPTR) DNS Resource Record
9. [RFC3261] SIP: Session Initiation Protocol
10. [RFC3262] Reliability of Provisional Responses in the Session Initiation Protocol(SIP)
11. [RFC3263] Session Initiation Protocol (SIP): Locating SIP Servers
12. [RFC3264] An Offer/Answer Model with the Session Description Protocol (SDP)
13. [RFC3311] The Session Initiation Protocol (SIP) UPDATE Method
14. [RFC3326] The Reason Header Field for the Session Initiation Protocol (SIP)
15. [RFC3550] RTP: A Transport Protocol for Real-Time Applications
16. [RFC3551] RTP Profile for Audio and Video Conferences with Minimal Control
17. [RFC3611] RTP Control Protocol Extended Reports (RTCP XR)
18. [RFC3960] Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
19. [RFC4566] SDP: Session Description Protocol
20. [RFC4733] RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
21. [RFC6337] Session Initiation Protocol (SIP) Usage of the Offer/Answer Model

Informative References

1. [RFC3312] Integration of Resource Management and Session Initiation Protocol (SIP)
2. [RFC3323] A Privacy Mechanism for the Session Initiation Protocol (SIP)
3. [RFC3325] Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
4. [RFC3329] Security Mechanism Agreement for the Session Initiation Protocol (SIP)
5. [RFC4032] Update to the Session Initiation Protocol (SIP) Preconditions Framework
6. [T.38] ITU-T, T.38 : Procedures for real-time Group 3 facsimile communication over IP networks
7. https://www.excentis.com/blog/packetcable-2-0-edva-config-files/

PacketCable 2.0 terminology

- Remark: against the part 1, both operator and user terminology refer only to the voice services.

RST The PacketCable 2.0 Residential SIP Telephony (RST) model is an abstract representation of the capabilities and features available on a phone line. The intent is to provide flexibility for future services. The RST specification introduces the following terms:

Operator The operator indirectly associates users with a P-CSCF. Multiple operators could be associated with various third-party telephony providers or with remote locations in the MSO's network.

User A user consists of two components:

• **IMPU (IP Multimedia Public User)** — the public identity of the user. This could be the phone number assigned to the user, or some other unique username. The IMPU associates a user with a phone line through the **pktcEUEUsrIMPUAdditionalInfo** object. Multiple users may share a line, implementing a Teen Line service or as part of a transition to a new area code or exchange.

• **IMPI (IP Multimedia Private Identity)** — the private identity of the user. Primarily, the IMPI defines the credentials for the IMPU.

Application Defines the service(s) to be provided to a user. The application map defines the application and selects a profile.

Profile Selects a digit map and an associated collection of features. A profile may be applied to one or more users.

Feature Defines capabilities and dialing features available to a profile. Features include basic call functionalities.

Voice Platform

VOO is using currently the ZTE IMS solution. SIP UE can access IMS via a SBC running version V5.22.36.P3 , such platform can be subject to change in future.

There is no COPS Gate control implemented. Call tapping (CALEA) can be performed by ZTE IMS solution or by the CISCO CBR8 CMTS (cable modem termination systems).

MTA connectivity

General

- The embedded MTA e-(UE) unit must be Euro packet cable 2.0 compliant.
- The e-UE must use DHCP option 60 in DHCP Discover messages to advertise PacketCable support. The option contains the string “pktc2.0” to indicate PacketCable 2.0 support.
- The provisioning method is BASIC.1
- The DHCP option 122.1 on the CM DHCP Server will trigger MTA start.
- DQOS is only active by DSx from Off/On-hook actions. (FXS port)
- The e-UE MAC = HFC MAC +1 by default.
- The ToD server (Time of the Day Server) is implemented on each CISCO CBR8 CMTS (Global Command: “**cable time server**”)
- PacketCable Authorization is disabled on CISCO CBR8 CMTS (we use global command:

“packetcable authorize vanilla-docsis-mta”)

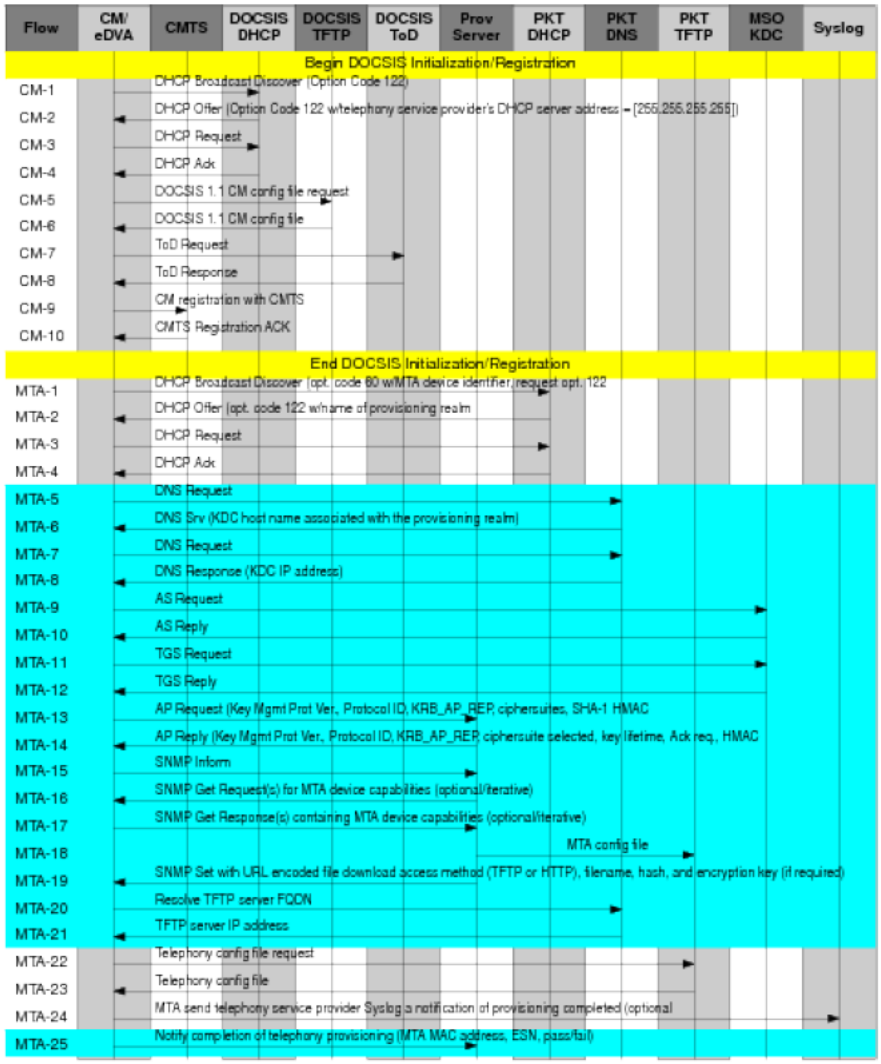
- Specific service flows and classifiers must be defined in the CM config file for SIP signaling & DSX request (DQoS) (see below)

e-UE (MTA) specific

- The SIP UE MUST use a dedicated IP interface that is specifically provisioned as the Voice Service [Voice Interface].
- The SIP UE MUST obtain an IP address using standard DHCP [RFC2131].
- The SIP UE MUST use IPv4.
- The SIP UE MUST NOT be used behind a NAT.
- The SIP UE MUST NOT use IPv6.
- The SIP UE MUST announce itself as a Packetcable device by containing string “pktc2.0” in DHCP option 60 (= Vendor Class Id)
- The SIP UE MUST at least request following options (Parameter Request List) to the DHCP Server (option 55 list):
    - 1 = Subnet Mask
    - 3 = Router,
    - 6 = Domain Name Server.
- The SIP UE MUST NOT be configured with a static IP address.
- The SIP UE MUST conform to the requirements in DNS standards: [RFC1034], [RFC1035],[RFC2782], [RFC2915].
- The SIP UE MUST follow standard industry best practice behavior with regards to usage of TTL.

e-UE (MTA) provisioning Mode

VOO uses BASIC.1 Provisioning flow. There is no (Kerberos) real name nor SNMPv3 (Secure Packet Cable Mode), there is no SNMP inform when the provisioning is complete (option.1). The steps in light blue are skipped from the complete set of provisioning steps.



The following parameters will be given by DHCP during the e-UE DHCP DORA transaction:

DHCP Options	Description	Value
Option 1	Subnet Mask	
Option 2	Time Offset	60 min
Option 3	Router (Def. Gw.)	
Option 4	Time Server (ToD)	= local CMTS
Option 6	Domain Name Server(s)	"10.231.1.98,10.231.65.98"
Option 7	Syslog Server	(Optional) 10.11.0.22
Option 15	Domain Name	voip.voo.be
Option 28	e-UE IP address	
Option 51	Lease Time	1 week

Option 66	Next Server (=Tftp Boot Server)	10.11.10.230
Option 122 CableLabs Client Configuration: 122.1 will be also present on CM DORA[1] transaction to trigger the e-UE Voip start 122.1 & 2 are specific per area (4)		
Option 122.1	Primary-dhcp-server	10.11.0.14 *(per area)*
Option 122.2	Secondary-dhcp-server	10.11.0.15 *(per area)*
Option 122.3	Provisioning server	ns1.voip.voo.be
Option 122.6	Kerberos-realm	"BASIC.1"

Remark: in PacketCable 2.0 (3GPP minded), the P-CSCCF: Proxy-Call Session Control Function, communicate with the SIP registrar. There is no need to provide it.

CM Configuration File Add-on

As noted earlier, specific voice signalling (SIP & DSX) service flow & classifiers must be created in the CM configuration file as follows:

Remark: (we use formalism from DOCSIS Linux editor):

```
/* Downstream IP */

DsServiceFlow
    {
        DsServiceFlowRef 8;
        ServiceClassName "VoIP_SIG_DS";
        QosParamSetType 7;
    }
    DsPacketClass
    {
        ClassifierRef 62;
        ServiceFlowRef 8;
        ActivationState 1;
        RulePriority 16;
        IpPacketClassifier
        {
            IpProto 256;
```

[1] DORA: DHCP Transaction: **D**iscover-**O**ffer-**R**equest-**A**ck

```
            DstPortStart 5060;  /* SIP protocol Port Number */
            DstPortEnd 5060;   /* SIP protocol Port Number */
        }
    }

 /* Upstream SIP & DSX */

    UsServiceFlow
    {
        UsServiceFlowRef 7;
        ServiceClassName "VoIP_SIG_US";
        QosParamSetType 7;
    }
    UsPacketClass
    {
        ClassifierRef 61;
        ServiceFlowRef 7;
        ActivationState 1;
        RulePriority 16;
        IpPacketClassifier
        {
            IpProto 256;
            SrcPortStart 5060; /* SIP protocol Port Number */
            SrcPortEnd 5060;  /* SIP protocol Port Number */
        }
    }
    UsPacketClass
    {
        ClassifierRef 60;
        ServiceFlowRef 7;
        ActivationState 1;
        RulePriority 16;
        LLCPacketClassifier
        {
            EtherType 0x030f16;  /* DSx Packet */
```

```
        }
    }
```

The service class names are defined in each CMTS for uniformity purpose:

cable service class 201 name ***VoIP_SIG_DS***
cable service class 201 downstream
cable service class 201 max-rate 135680
cable service class 201 max-burst 4000
cable service class 201 priority 4

cable service class 202 name ***VoIP_SIG_US***
cable service class 202 upstream
cable service class 202 sched-type 2
cable service class 202 tos-overwrite 0 68
cable service class 202 max-rate 143360
cable service class 202 min-rate 0
cable service class 202 priority 4

MTA Profile

Configuration of SIP client

VOO will provide a set of SIP credentials to the end user, one set for each assigned phone

number. The following entries MUST be configurable in the SIP UE (separately for each assigned phone number):
a. Phone Number (below referred to as “PHONE_NUMBER”)
b. SIP Domain (below referred to as “SIP_DOMAIN”)
c. Outbound SIP Proxy
d. (Authentication) Username
e. (Authentication) Password

The SIP UE MUST support a Username in the following format:
“PHONE_NUMBER@SIP_DOMAIN”
The SIP UE MUST support a password in the format of this regular expression:
    “[a-zA-Z0-9!$/()=?*+#-_.:,]{30,40}”

Standard SIP Support

The SIP UE MUST be compliant to the base SIP specification [RFC3261], implementing the "User Agent (UA) role" in particular.

The SIP UE MUST support DNS SRV [RFC2782] and DNS A [RFC1035] record queries for locating the SIP server as defined in [RFC3263]. These are to be used to resolve the provided Outbound Proxy into an IP address, destination port and protocol to be used.
The SIP UE MUST use the SIP port as provided in the DNS SRV [RFC2782] response as defined in [RFC3263] to contact its outbound proxy.
The Public User Identity (IMPU) MUST take the form of a SIP URI as specified in [RFC3261].
The Public User Identity (IMPU) MUST be built as follows (based on the phone-number and SIP Domain): sip:PHONE_NUMBER@SIP_DOMAIN
Any other URI format (e.g., tel URI) MUST NOT be used by the SIP UE.
The SIP UE MUST never put "anonymous" in any outgoing message.
The SIP UE MUST be compliant to [RFC3262] (Provisional Acknowledgement - PRACK).
The SIP UE MUST be compliant to [RFC3311] (aka. SIP-UPDATE).
The SIP UE MUST be compliant to [RFC3960] (Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)).
The SIP UE MUST be compliant to [RFC6337] (Session Initiation Protocol (SIP) Usage of the Offer/Answer Model).
The SIP UE MUST NOT use the Privacy mechanism for SIP specified in [RFC3323]
The SIP UE MUST NOT use the Private Extensions to SIP specified in [RFC3325].
The SIP UE MUST be compliant to [RFC3326] (Reason Header Field).

SIP Registration and Redundancy

The SIP UE MUST be compliant to [RFC3263] (Locating SIP Servers). (see above)
The SIP UE MUST support DNS SRV [RFC2782] and DNS A [RFC1035] record queries for locating the SIP server as defined in [RFC3263]. (see above) .
These are to be used to resolve the provided Outbound Proxy into an IP address, destination port and protocol to be used .If more than one SIP servers are resolved, the SIP UE MUST always try to register to the priority SIP server coming from DNS SRV response. The second priority SIP server is only used in case of an outage of the first priority SIP server.
The SIP UE MUST NOT register against two SIP servers in parallel.
In case of a SIP UE failover and registration to the second priority SIP server, the SIP UE MUST try with the next re-registration attempt to register all phone numbers back to the first priority SIP server again.
If the SIP UE re-registration attempt towards first priority SIP server fails, the SIP UE MUST

stay registered on the second priority SIP server but the fallback to the first priority SIP server MUST be retried again with the next re-registration attempt.
If SIP UE fallback registration attempt to the first priority SIP server is successful for one phone number all other phone numbers MUST fallback too.
If the IP address of the SIP UE changes (e.g., upon DHCP Renew), the SIP UE MUST start a new SIP registration.

SIP Security

The SIP Digest method as specified in [RFC3261] MUST be supported. Any other mechanisms or protocols to protect the SIP signalling MUST NOT be used. On receiving a “401 (Unauthorized)” response to the REGISTER request, and where the "algorithm" Authorization header field parameter is "MD5", the SIP UE MUST extract the digest-challenge parameters as indicated in [RFC2617] from the WWW-Authenticate header field, calculate digest-response parameters as indicated in [RFC2617], send another REGISTER request containing an Authorization header field containing challenge response as indicated in [RFC2617]. The SIP UE MUST set the Call-ID of the REGISTER request which carries the authentication challenge response to the same value as the Call-ID of the 401 (Unauthorized) response which carried the challenge. The SIP UE MUST NOT include [RFC3329] header fields with this REGISTER. The SIP UE SHOULD only accept SIP requests from the SIP server IP addresses that are resolved via DNS SRV / A records (SIP server whitelist).

SIP Sessions – Originating

A SIP INVITE message MUST only be sent out by SIP UE during an active registration and only to the SIP server on which the SIP UE is currently registered. The SIP UE as Caller (side A) MUST NOT apply local ring back tone towards the connected telephone set when it receives “180 (Ringing)” with SDP. The SIP UE as Caller (side A) MUST NOT apply local ring back tone towards the connected telephone set when it receives “183 (session progress)” with SDP. An outgoing INVITE MUST use the following format for the “Request-URI” field: sip:CALLED_PHONE_NUMBER@SIP_DOMAIN

An outgoing INVITE MUST use the following format for the “To:” field:
<sip:CALLED_PHONE_NUMBER@SIP_DOMAIN>

Please check Annex 1 part for digit map details

SIP Sessions – Terminating

When it receives an INVITE the SIP UE as Callee (side B) MUST respond with “180 (Ringing)” without SDP.

SDP Profile

The SIP UE MUST be compliant to [RFC3264] (Offer/Answer Model with SDP). The SIP UE MUST be compliant to [RFC4566] (Session Description Protocol). An INVITE request generated by a SIP UE MUST contain an SDP offer with at least one media description. This SDP offer MUST reflect the calling user's terminal capabilities and user preferences for the session. The SIP UE MUST NOT request/enable authentication/encryption for the media streams. Upon sending an SDP answer to an SDP offer (which included one or more media lines which was offered with several codecs) the terminating SIP UE MUST select exactly one codec per media line and indicate only the selected codec for the related media stream. The SIP UE MUST support configuration of dynamic RTP payload type number which is used for DTMF RTP Events as defined by [RFC4733]. The SIP UE MUST be configured to use DTMF RTP Events as defined by [RFC4733], then it MUST add the following in its outgoing SDP:

a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

The SIP UE MUST include the "a=ptime" attribute for all "audio" media lines as described in [RFC4566], with value 20. If a SIP UE receives an "audio" media line with "a=ptime" specified, the SIP UE MUST transmit at the specified packetization rate. If the SIP UE supports RTCP Extended Reports per [RFC3611], then this MUST be indicated in its SDP offer per [RFC3611] with encoding “a=rtcp-xr:voip-metrics”. The SIP UE MUST NOT try to set up any video sessions.

MTA Generic Configuration File

Binary file is part of the Pack: Standard_config.bin

Remark:

ClabProjPacketCable = .1.3.6.4.1.4491.2.2.

Please refer to:

- CL-PKTC-EUE-DEV-MIB/
  .1.3.6.1.4.1.4491.2.2.**10.3**
- CL-PKTC-EUE-USER-MIB/
  .1.3.6.1.4.1.4491.2.2.**10.4**
- CL-PKTC-EUE-RST-MIB/
  .1.3.6.1.4.1.4491.2.2.**8.**
- CL-PKTC-EUE-RST-MIB/#pktcEUERSTAppFeatID

.1.3.6.1.4.1.4491.2.2.**8.2**

- PKTC-SIG-MIB/

  .1.3.6.1.4.1.4491.2.2.**2.**
- PKTC-IETF-MTA-MIB/

  .1.3.6.1.2.1.**140**

Config file permit to fulfill some packetCable defined tables.

Main	Info Tables :
{	
MtaConfigDelimiter 1;	Start file
SnmpMibObject mib-2.140.1.1.6.0 Integer 1 ;	PktcMtaDevEnabled
/* Op Table */	
SnmpMibObject clabProjPacketCable.**10.3.1.1.2**.1.2.1 String "voip.voo.be" ;	Table/Operator Domain
SnmpMibObject clabProjPacketCable.10.3.1.1.2.1.11.1 Integer 4 ;	Create & Go
/*PCSCF Table */	
SnmpMibObject clabProjPacketCable.**10.3.1.1.4.**1.2.1.1 Integer 16 ;	PCSCFAddrType = ipv4
SnmpMibObject clabProjPacketCable.10.3.1.1.4.1.3.1.1 String "sipregistrar.voip.voo.be" ;	PCSCFAddr
SnmpMibObject clabProjPacketCable.10.3.1.1.4.1.4.1.1 Gauge32 5060 ;	PCSCFSipPort
SnmpMibObject clabProjPacketCable.10.3.1.1.4.1.12.1.1 Integer 4 ;	Create & Go
/*IMPU (Public User) Table */	
SnmpMibObject clabProjPacketCable.**10.4.1.1.2**.1.2.1 Integer 1 ;	IMPU index
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.3.1 String "+3272809941" ;	IMPU Id
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.4.1 Gauge32 1 ;	IMPI Index
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.6.1 String "1" ;	IMPU Operator Index
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.11.1 Integer 2 ;	IMPU Sig Security
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.13.1 Integer 4 ;	Create & Go
/*IMPI (Private Identity) Table */	
SnmpMibObject clabProjPacketCable.**10.4.1.1.3.**1.2.1 Integer 1 ;	IMPI ID Type
SnmpMibObject clabProjPacketCable.10.4.1.1.3.1.3.1 String "+3272809941@voip.voo.be" ;	IMPI ID
SnmpMibObject clabProjPacketCable.10.4.1.1.3.1.4.1 Integer 3 ;	IMPI Credentials Type
SnmpMibObject clabProjPacketCable.10.4.1.1.3.1.5.1 String "abCDefGHij" ;	IMPI Credentials
SnmpMibObject clabProjPacketCable.10.4.1.1.3.1.6.1 Integer 4 ;	Create & Go
/*Digit Map Features Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.1.3**.1.1.3.1 Integer 4 ;	Create & Go
/*Application Profile to Features Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.1.2**.1.3.1.2 Integer **2** ;	App Feature ID
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.2 Gauge32 1 ;	App Feature Index Reference
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.2 Integer 4 ;	Create & Go
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.3 Integer **3** ;	....
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.3 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.3 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.5 Integer **5** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.5 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.5 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.6 Integer **6** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.6 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.6 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.6 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.7 Integer **7** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.7 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.7 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.7 Integer 4 ;	

SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.8 Integer **8** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.8 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.8 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.8 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.10 Integer **10** ;	/*APP Feature ID4, 9 & 11 not used*/
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.10 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.10 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.10 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.12 Integer **12** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.12 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.12 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.12 Integer 4 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.3.1.13 Integer **13** ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.4.1.13 Gauge32 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.5.1.13 Integer 1 ;	
SnmpMibObject clabProjPacketCable.8.2.1.1.2.1.9.1.13 Integer 4 ;	
SnmpMibObject clabProjPacketCable.**8.2.1.2.6.**1.1.5.1 Integer 4 ;	Create & Go Caller ID Display
/* Standard Basic Call Features */	Feature Status
/* Standard Basic Call Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.1.1**.1.2.1 String "PCMA" ;	Preferred Codec List
SnmpMibObject clabProjPacketCable.8.2.1.2.1.1.1.3.1 Integer 4 ;	Create & Go
/*Standard Network Forwarding (Nf) Basic Call Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.1.2**.1.1.1 Gauge32 1 ;	Call Bye Delay
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.2.1 Gauge32 30 ;	Dial Tone Timer
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.4.1 Gauge32 10 ;	Termination Error Timerr
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.6.1 Gauge32 10 ;	Perm. Seq. Timer 1
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.8.1 Gauge32 10 ;	Perm.Seq.Timer 2
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.10.1 Gauge32 10 ;	Perm.Seq.Timer 3
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.11.1 Gauge32 20 ;	Lockout Reset Timer
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.12.1 Gauge32 15 ;	RTP Non-emerg.pckt DSCP
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.13.1 Gauge32 15 ;	RTCP Sig. DSCP
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.14.1 Integer 4 ;	Create & GO
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.15.1 Gauge32 16 ;	InterDigit Timer
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.17.1 Gauge32 10 ;	Perm.Seq.Timer 4
SnmpMibObject clabProjPacketCable.8.2.1.2.1.2.1.18.1 Integer 2 ;	No Override of reject
/* Res.Serv.Tel (RST) Active Status Change Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.3.1**.1.2.1 Gauge32 3600 ;	Registration Expiration
SnmpMibObject clabProjPacketCable.8.2.1.2.3.1.1.3.1 Integer 4 ;	Create & GO
/* Res.Serv.Tel (RST) No answer Timeout Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.4.**1.1.2.1 Gauge32 180 ;	No Answer Timeout (TO)
SnmpMibObject clabProjPacketCable.8.2.1.2.4.1.1.3.1 Integer 4 ;	Create & GO
/* Res.Serv.Tel (RST) Caller ID Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.5.**1.1.2.1 Integer 2 ;	No Permanent Presentation
SnmpMibObject clabProjPacketCable.8.2.1.2.5.1.1.3.1 Integer 4 ;	Create & GO
/* Res.Serv.Tel (RST) Call DeliveryTable */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.8.**1.1.2.1 String "file:///PacketCableRST/cf" ;	Confirmation Tone
SnmpMibObject clabProjPacketCable.8.2.1.2.8.1.1.3.1 String "file:///PacketCableRST/ro" ;	Error Tone
SnmpMibObject clabProjPacketCable.8.2.1.2.8.1.1.4.1 Integer 4 ;	Create & GO
/* Res.Serv.Tel (RST) Call Wait Table */	
SnmpMibObject clabProjPacketCable.**8.2.1.2.10**.1.1.2.1 Integer 1 ;	Call Wait Cancel hook flash
SnmpMibObject clabProjPacketCable.8.2.1.2.10.1.1.3.1 Integer 4 ;	Create & GO
/*IMPU Table */	
SnmpMibObject clabProjPacketCable.10.4.1.1.2.1.12.1 String "IEP#9 OEP#9" ;	Additional Info per PcktCable
SnmpMibObject clabProjPacketCable.10.4.1.1.4.1.2.1.1 Gauge32 4491 ;	Organization ID (CL:4491)
SnmpMibObject clabProjPacketCable.10.4.1.1.4.1.3.1.1 Gauge32 1 ;	App ID
SnmpMibObject clabProjPacketCable.10.4.1.1.4.1.4.1.1 Gauge32 1 ;	App Index Ref.

SnmpMibObject clabProjPacketCable.10.4.1.1.4.1.9.1.1 Integer 4 ;	Create & GO
/* Signalling Endpoint Configuration Table (in millisec)*/	
SnmpMibObject pktcSigEndPntConfigMinHookFlash.9 Gauge32 60 ;	Min Hook Flash line1 (ms)
SnmpMibObject pktcSigEndPntConfigMinHookFlash.10 Gauge32 60 ;	Min Hook Flash line 2 (ms)
SnmpMibObject pktcSigEndPntConfigPulseDialMinMakeTime.9 Gauge32 25 ;	Min Pulse Width line 1 (ms)
SnmpMibObject pktcSigEndPntConfigPulseDialMinMakeTime.10 Gauge32 25 ;	Min Pulse Width line 2 (ms)
SnmpMibObject pktcSigEndPntConfigPulseDialMaxMakeTime.9 Gauge32 55 ;	Max Pulse Width line 1 (ms)
SnmpMibObject pktcSigEndPntConfigPulseDialMaxMakeTime.10 Gauge32 55 ;	Max Pulse Width line 2 (ms)
SnmpMibObject pktcSigEndPntConfigPulseDialMinBreakTime.9 Gauge32 45 ;	Min Break pulse width line 1
SnmpMibObject pktcSigEndPntConfigPulseDialMinBreakTime.10 Gauge32 45 ;	Min Break pulse width line 2
SnmpMibObject pktcSigEndPntConfigPulseDialMaxBreakTime.9 Gauge32 75 ;	Max Break pulse width line 1
SnmpMibObject pktcSigEndPntConfigPulseDialMaxBreakTime.10 Gauge32 75 ;	Max Break pulse width line 2
SnmpMibObject ifAdminStatus.9 Integer 1; /* up */	Line 1 activation
SnmpMibObject ifAdminStatus.10 Integer 2; /* down */	Line 2, no activation
SnmpMibObject clabProjPacketCable.8.2.1.1.3.1.1.2.1 HexString 0x2f2f2054696d65722076616c7565730d0a54494d4552205a3d322e30303030303030202f2f53686f727 420496e746572646967697420546 96d65720d0a54494d455220533d342e30303030303030202f2f4c6f6 e6720447572617469 6f6e2054696d6572200d0a0d0a2f2f2053796d626f6c730d0a646f6d61696e203d 202240766f69702e766f6f2e6265220d0a0d0a2f2f204d6170730d0a4d4150204d61696e5461626c652 03d0d0a2022283d456d657267656e63794e756d6265722922203a204d414b452d43414c4c2822736 9703a2220233176203d646f6d61696e290d0a2022283d4e6174696f6e616c4e756d6265722922203a 204d414b452d43414c4c28227369703a2220233176203d646f6d61696e290d0a2022283d496e744e7 56d6265722922203a204d414b452d43414c4c28227369703a2220233176203d646f6d61696e290d0a 2022283d566572746963616c536572766963 65436f64652922203a2052455455524e0d0a2022283d5 3686f72744e756d6265722922203a204d414b452d43414c4c28227369703a2220233176203d646f6d 61696e290d0a0d0a2f2f204d617463686 57320456d657267656e6379204e756d62657273200d0a4d4 15020456d657267656e63794e756d626572203d20200d0a20222831307829283d656e642922203a2 052455455524e282331292020200d0a2022283131782928 ;	Digit Map string Value (ABNF)
SnmpMibObject mib-2.140.1.2.11.0 HexString 0x4de71e629d6adfc43d5a5416ddf00a3f7b97d1c1 ;	Hash value of the content
MtaConfigDelimiter 255;	End of file
}	

Configure the Application Map

Cf. Annex 1

MTA SIP Signalling

REGISTER

```
REGISTER sip:SIP_DOMAIN SIP/2.0
Via: SIP/2.0/UDP UE_ADDRESS;branch=aaaabbbbcccc;rport
Route : <sip: Outbound SIP Proxy. SIP_DOMAIN:port;lr>
Max-Forwards: 70
From: <sip:PHONE_NUMBER@SIP_DOMAIN>;tag=123456789
To: <sip:PHONE_NUMBER@SIP_DOMAIN>
Call-ID: xxxxyyyyzzzz
CSeq: 123 REGISTER
```

```
Allow: ACK,BYE,CANCEL,INVITE, NOTIFY, OPTIONS, PRACK, REFER, SUBSCRIBE, UPDATE
Allow-Events: dialog, message-summary, refer, reg, ua-profile
Authorization: Digest
username="AUTH_USER", realm="SIP_REALM", nonce="NONCE_VALUE",
uri="sip:SIP_DOMAIN", response="AUTH_RESPONSE", algorithm=MD5,
opaque="aaaaaAAAAAAAAA==",qop=auth,cnonce="aaBB0097",nc=00000001
Contact: <sip:PHONE_NUMBER@ SIP_DOMAIN :port>
Supported: 100rel, answermode, early-session, eventlist, from-change, histinfo, join, outbound, path, replaces,
tdialog, timer
User-Agent: UE_DETAILS
Content-Length: 0
```

INVITE

```
INVITE sip:CALLED_PHONE_NUMBER@SIP_DOMAIN SIP/2.0
Accept: application/reginfo+xml, application/rlmi+xml, application/sdp, application/server-call-feature-status+xml,
application/simple-message-summary, application/watcherinfo+xml, application/xml, message/sipfrag,
multipart/mixed
Via: SIP/2.0/UDP UE_ADDRESS;branch=aaaabbbbcccc;rport
Route : <sip: Outbound SIP Proxy. SIP_DOMAIN:port;lr>
Max-Forwards: 70
From: <sip:PHONE_NUMBER@SIP_DOMAIN>;tag=123456789
To: <sip:CALLED_PHONE_NUMBER@SIP_DOMAIN>
Call-ID: xxxxyyyyzzzz
CSeq: 123 INVITE
Allow: ACK,BYE,CANCEL,INVITE, NOTIFY, OPTIONS, PRACK, REFER, SUBSCRIBE, UPDATE
Allow-Events: dialog, message-summary, refer, reg, ua-profile
Contact: <sip:PHONE_NUMBER@ SIP_DOMAIN :port>
Min-SE: 90
P-Preferred-Identity: sip:PHONE_NUMBER@SIP_DOMAIN
Session-Expires: SESSION_EXPIRATION_DELAY
Supported: 100rel, answermode, early-session, eventlist, histinfo, join, outbound, path, replaces, tdialog, timer
User-Agent: UE_DETAILS
P-Early-Media: supported
Content-Disposition: session
Content-Type: application/sdp
Content-Length: 369
v=0
o=<username> <sess-id> <sess-version> IN IP4 <unicast-address>
s=-
c=IN IP4 <connection-address>
t=0 0
m=audio <port> RTP/AVP 8 101
b=TIAS:64000
b=AS:80
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=rtcp:54457
```

```
a=sendrecv
a=silenceSupp:on - - - -
a=maxprate:50.0
a=rtcp-xr:voip-metrics
```

Call features / Supplementary Services

The SIP UE MUST NOT interpret Vertical Service Codes (VSC) locally.

VSCs follow this regular expression: ^(*67){0,1}*[0-9]{1,25}$

Overview of VSCs: The e-UE **must comply** with the mini set of Service VSC below:

ID	Description	Status	Comments
1	Calling Line Identification Restriction *31*<target phone number>#	Y-N	
2	Calling Line Identification Presentation per call *32*<target phone number>#		
3	Call Forwarding Unconditional – Activation *21*<target phone number>#		
4	Call Forwarding Unconditional – Deactivation #21#		
5	Call Forwarding Busy – Activation *67*<target phone number>#		
6	Call Forwarding Busy – Deactivation #67#		
7	Call Forwarding No Reply – Activation *61*<target phone number>#		
8	Call Forwarding No Reply – Deactivation #61#		
9	VSCs beginning with an asterisk (*) MUST be sent transparently and unchanged by the SIP		
10	VSCs beginning with an asterisk (*) MUST be sent transparently and unchanged by the SIP		
11	UE to the SIP server (per the digit map as specified below in Annex part).		
12	VSCs beginning with a hash (#) MUST be sent transparently and unchanged by the SIP UE to the SIP server (per the digit map in Annex 1)		
13	Double line (line 1 to line 2)		

14	Double line (line 2 to line 1)		

Assistance

Should you need some assistance, you can contact Orange/VOO to know the conditions: cable.engineering@orange.be

This email address is subject to change without prior notice, please check our website to get our latest version.

Annex 1 Digit-map

Digit-Map

RST Digit Map Index - 1

// Timer values

TIMER Z=2.000000 //Short Interdigit Timer

TIMER S=4.000000 //Long Duration Timer

// Symbols

domain = "@voip.voo.be"

// Maps

MAP MainTable =

"(=EmergencyNumber)" : MAKE-CALL("sip:" #1v =domain)

"(=NationalNumber)" : MAKE-CALL("sip:" #1v =domain)

"(=IntNumber)" : MAKE-CALL("sip:" #1v =domain)

"(=VerticalServiceCode)" : RETURN

"(=ShortNumber)" : MAKE-CALL("sip:" #1v =domain)

// Matches Emergency Numbers

MAP EmergencyNumber =

"(10x)(=end)" : RETURN(#1)

"(11x)(=end)" : RETURN(#1)

```
// Matches Dialed Number

MAP NationalNumber =

"(01xxxxxxx)(=end)" : RETURN(#1)

"(02xxxxxxx)(=end)" : RETURN(#1)

"(03xxxxxxx)(=end)" : RETURN(#1)

"(04xxxxxxx)(=end)" : RETURN(#1)

"(05xxxxxxx)(=end)" : RETURN(#1)

"(06xxxxxxx)(=end)" : RETURN(#1)

"(07xxxxxxx)(=end)" : RETURN(#1)

"(08xxxxxxx)(=end)" : RETURN(#1)

"(09xxxxxxx)(=end)" : RETURN(#1)

"(077xxxxxxxxxxx)(=end)" : RETURN(#1)

"(04xxxxxxxx)(=end)" : RETURN(#1)


// Matches Dialed Number

MAP IntNumber =

"(00[0-9]{1-18})(=End)" : RETURN(#1)


// Matches Vertical Service Codes

MAP VerticalServiceCode =

  "(*21*x{1-22}#)"     : MAKE-CALL ("sip:" #1 =domain) //CALL_FORWARD_ENABLE

  "(*31*x{1-22}#)"     : MAKE-CALL ("sip:" #1 =domain) //CALLER_ID_BLOCK_PER_CALL_ENABLE

  "(*32*x{1-22}#)"     : MAKE-CALL ("sip:" #1 =domain) //CALLER_ID_BLOCK_PER_CALL_DISABLE

  "(*61*x{1-22}#)"     : MAKE-CALL ("sip:" #1 =domain)
//CALL_FORWARD_NO_ANSWER_ENABLE
```

```
"(*67*x{1-22}#)"    : MAKE-CALL ("sip:" #1 =domain) //CALL_FORWARD_BUSY_ENABLE
"*43*"    : MAKE-CALL("sip:*43*" =domain) //CANCEL_CALL_WAITING
"*20*"    : MAKE-CALL("sip:*20*" =domain) //CALLER_ID_DELIVERY_BLOCKING_ENABLE
"*37*"    : MAKE-CALL("sip:*37*" =domain) //ACTIVATE_BUSY_LINE_AUTO_CALLBACK
"*80*"    : MAKE-CALL("sip:*80*" =domain) //WAKE_UP_SERVICE_ENABLE
"*98*"    : MAKE-CALL("sip:*98*" =domain) //CALL_BAR_CHANGE_PIN
"*99*"    : MAKE-CALL("sip:*99*" =domain) //CALL_BAR_DISABLE
"#21#"    : MAKE-CALL("sip:#21#" =domain) //CALL_FORWARD_DISABLE
"#61#"    : MAKE-CALL("sip:#61#" =domain) //CALL_FORWARD_NO_ANSWER_DISABLE
"#67#"    : MAKE-CALL("sip:#67#" =domain) //WAKE_UP_SERVICE_DISABLE
"#80#"    : MAKE-CALL("sip:#80#" =domain) //CALL_BAR_ENABLE
"#99#"    : MAKE-CALL("sip:#99#" =domain) //CALL_FORWARD_BACKUP_ENABLE


// Matches Short Numbers
MAP ShortNumber =
"(1x{3})(=end)" : RETURN(#1)
"(1x{5})(=end)" : RETURN(#1)


// Matches End of Dial Indicators # & S
MAP End =
"#" : RETURN
"S" : RETURN
```