Changing the Way the World Communicates

IPNx Sat ESA Project
Final Report

World Telecom Labs

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1. Version History

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Revision	Date	Author	Description
0.1	13/12/07	SP	First revision.
1.0	18/12/07	SP	Published Version

2. Introduction

2.1 Document Purpose

This document summarises the work carried out in WTL's ESA funded Artes 4 project: "**VoIP Equipment for Satellite Use"** Contract 19344.

2.2 Intended Audience

This document is intended for publication. It should provide useful information for WTL current and prospective customers, ESA's partners, and technical personnel who are interested in the state of the art in VoIP over satellite.

2.3 Related Documentation and Links

For more information on WTL products, please see www.wtl.be or email sales@wtl.be

3. Contents

4. Summary

World Telecom Labs NV (WTL), a Brussels-based pioneer in VoIP equipment for telecom operators, has recently completed a development contract with the European Space Agency (ESA). The contract, under ESA's Artes 4 program, provided joint funding for a series of developments designed to make WTL equipment perform an even better job for telecom operators wishing to use emerging low cost satellite services.

WTL already has an excellent track record in providing equipment for voice services using satellite trunking. A key benefit in this area has always been the superior bandwidth saving capability of WTL's patented NOP (Network Optimisation Protocol). One focus of the ESA project was to modify NOP to operate efficiently over DVB-RCS services. The lower price point of DVB-RCS equipment and space segment means that this will be of great interest to operators, particularly in the developing world.

One of the main results of the project was to prove, using an independent voice quality laboratory, that WTL equipment matches or outperforms regular SIP and H.323 equipment whilst using <u>half</u> the satellite bandwidth.
This is believed to be the first project that ESA has funded in the area of VoIP, and indicates a firm belief in VoIP as a suitable technology for use over satellite, and in WTL equipment as a good platform for this.

5. Background

WTL supplies call switching equipment to the competitive telecom carrier market. Operators in this sector are normally more entrepreneurial, more cost-conscious, and less conservative than traditional, incumbent PTTs. There are currently more than 100 competitive carriers using WTL equipment in around 30 countries.

Typical WTL customers may be:

a) Retail operators,
b) Wholesale operators, or
c) Both

5.1 Retail Operators

As one might expect, Retail Operators sell calls direct to subscribers. This may be done in a combination of the following ways:

- Residential accounts
- Calling cards
- Call shops
- PC-based IP 'soft phone' clients
- VoIP phones

This business is 95% Pre-Paid and may be highly specialised. For example, the operator may offer very good services and low rates to certain limited destinations, or may perhaps target particular ethnic groups.

The Retail operator normally does not have their own routes. This means that they will not have their own infrastructure in the destination countries. Having collected the call from a subscriber they will seek to make money by Least Cost Routing (LCR) to the cheapest onward carrier for each destination.

The WTL equipment allows the operator to run such a service. A central real-time accounting database is available and is used as follows:

- Users call local access numbers.
  - Users can be fixed line, GSM, pay phone, PC phone, IP Phone, business or private.
- The user's calling number, IP address and/or PIN is checked to ensure they are a valid account holder.
- Destination number entered by user.
- User's current balance is queried.
  - Options to announce balance or time available to user.
- If enough balance is available, the call is routed by cheapest route.
- When the balance hits zero (or other limit), the call is ended.
  - Options to warn the caller before cutting call.

The WTL Pre-Paid systems are highly regarded and widely used in the telecoms market because of the following benefits:

- High reliability – allows the service to make money 24/7.
- High capacity – there are many installed systems with over 1million active cards/accounts.
- Flexibility – hundreds of service and charging options.
- Real Time Operation – balances reduce during the call.
- Security – excellent anti-fraud features.

5.2 Wholesale Operators

Typical Wholesale Operators sell call capacity in bulk to other operators. By definition, they will own infrastructure and end-to-end routes. However, these operators may offer just a single destination because they have a government license or good contacts in a country. As a rule the more "difficult" a destination, the more valuable it is, and the more the operator can charge per minute for delivering calls there.

It is important to note that these operators are likely to be very open-minded about the network used to deliver the calls. In the worst case this may mean VoIP over the public internet; in the best case it may entail reserved bandwidth on a submarine cable. The physical transmission will be 'whatever it takes' to reach the destination. Clearly, at times, satellite will be the only practical option, but the operator will not necessarily be a satellite expert, and so the choice of service may not be optimal. Also, WTL will frequently have no input to the type, capacity, or specification of the connection. The critical success factors for the wholesale business are

- Quality and reliability of the interconnection to the remote end operator.
- Quality & reliability of the transmission to and from the remote end.
- … But cost is also key.

This last point led WTL to develop a patented technology that allows the operator to use less bandwidth per VoIP call than standard equipment. This is called NOP (Network Optimisation Protocol), and it works by packing multiple voice samples into the same IP packet. As the following diagram shows, this contrasts with VoIP calls using H.323 or SIP (the standard protocols for VoIP).

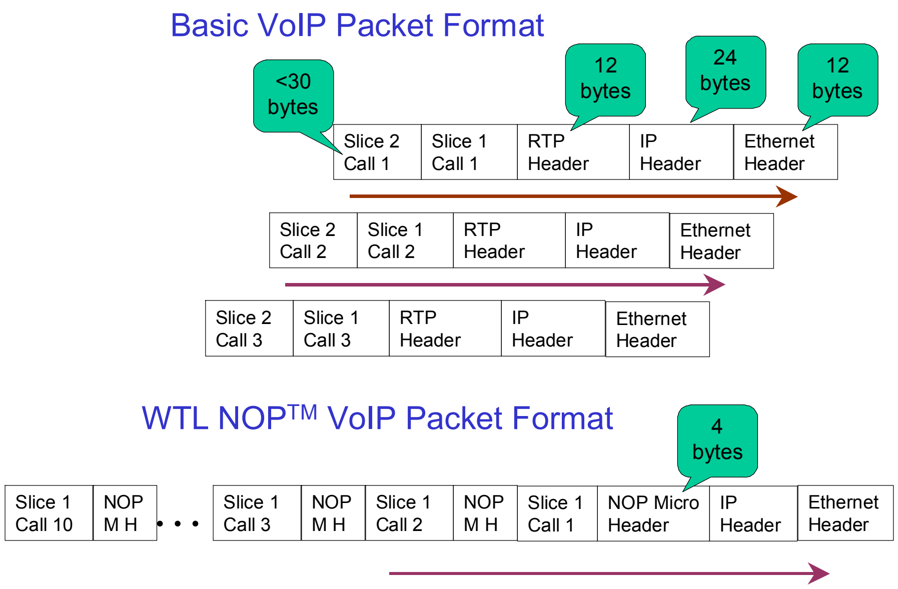

Figure 1 – NOP packets vs. basic VoIP

5.3 Description of the NOP Protocol

NOP works by assembling several voice packets that are going along the same network route into larger IP packets. This gives far better bandwidth utilization: the IP header of 40 bytes is replaced by a small 4 byte header which is inserted before each voice packet inside the large IP frame. This header identifies the type of packet (voice, silence, fax, DTMF), the size of the packet (a silence packet is smaller than a voice packet), and the destination channel.

Furthermore, we should note that combining several voice packets does not increase the transmission delay, as all these packets are generated at the same time by different calls. When 10 channels or more are combined, the IP overhead plus the 4 byte header on every voice packet (together with possible short silence frames), brings the overall bandwidth down to nearly 6.4 kbps per channel, the actual compression algorithm rate. This outstanding performance allows WTL to put 10 toll quality calls on one 64 kbps data channel while other products cannot support more than 5 or 6 calls. While doing this, the other products may suffer longer delays (because they wait to pack two voice samples from the same channel).
On a 100 call link, NOP will generate 25 times fewer IP packets than standard SIP or H.323.

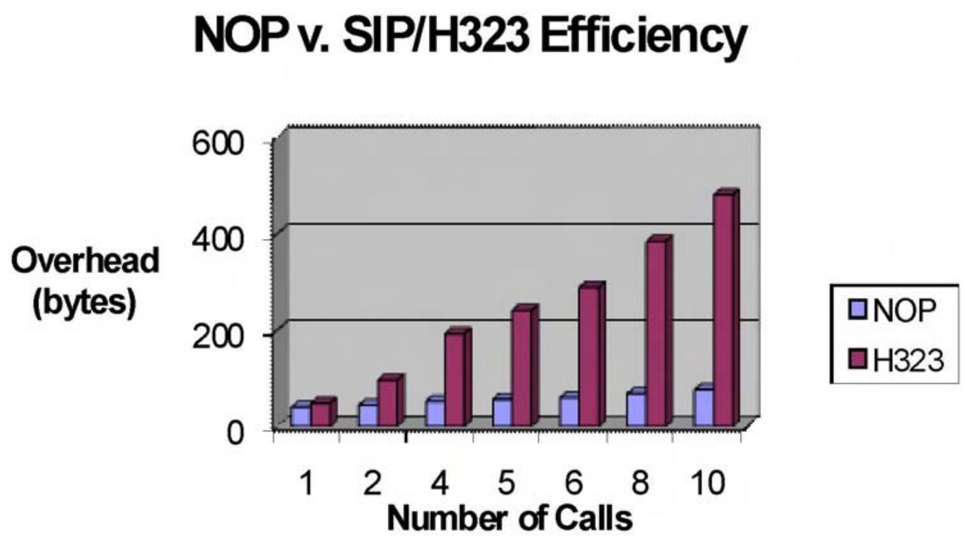

Figure 2 – NOP vs. SIP/H.323 Efficiency

Table 1 shows the Packets Per Second (PPS) generated by the different protocols: the more calls that NOP carries, the better it performs.

	Number of Calls					
	1	2	10	50	100	1000
SIP/H323	16.5 pps	33 pps	165 pps	825 pps	1650 pps	16,500 pps
NOP	33 pps	33 pps	33 pps	33 pps	66 pps	660 pps

Table 1 – Packets Per Second created by VoIP Protocols

6. Project Objectives

The project addressed the difficulties of carrying VoIP traffic over shared satellite services (especially DVB-RCS). NOP was already in use over a variety of satellite services, but there had been reports from customers of poorer than expected voice quality in some cases. The initial theory was that large jitter values and BoD (Bandwidth on Demand) in contention-based services were causing this, and so it was planned to adapt NOP to address this. A package of other improvements was also proposed at the same time in order to make the products more suitable for satcom operators.

The platform for the development was WTL's existing IPNx, SoIP and PVx switches, which are already successfully used by terrestrial telecoms operators all over the world. NOP is currently used to improve the bandwidth utilisation of VoIP traffic within a telephone network. The ultimate aim was that the resultant products would be suitable for use with VSAT or other low cost satellite services. The project included an extensive beta test in a live network, trialling the use of DVB-RCS links for VoIP.

The project was divided into 2 phases:

- Phase 1 involved constructing a DVB-RCS Reference Network; taking detailed performance measurements from it; and using emulation tools to characterise the issues currently affecting the quality of VoIP over satellite.
- Phase 2 involved implementing the improvements identified in Phase 1; using an independent voice quality testing laboratory to quantify these improvements; and finally testing the improvements in a Beta Test situation using VoIP over DVB-RCS with a live telecom operator.

7. Specific Tasks of the Project

7.1 Characterisation of VoIP over DVB-RCS

WTL are not satellite experts, despite the fact that WTL equipment is regularly connected via satellite networks. Generally, customers choose their own physical links.
Fundamental to this project was the aim of improving the performance of specialist WTL protocols within a satellite network environment, and specifically with DVB-RCS links. Therefore, WTL engineers needed to be able to send different types of traffic (NOP, remote management, “Soft IVR”) over a DVB-RCS link and record the results.
The Belgacom Sky 3 service was used for this testing, with a maximum downstream link of 1024 kbps and 256 kbps upstream. Naturally, since VoIP is a symmetrical data flow, this service was constrained by the lower capacity link: 256 kbps in this case. The quoted capacities are maximum figures. The worst case performance on a fully contended service was 25% of the maximum. This would equate to 64 kbps which, with the use of NOP, should be sufficient for 8 to 10 calls (depending on the voice codec used and other settings). One way delay was approximately 250ms, but jitter may be as high as 100ms.

It was clear from the tests carried out that it would not be possible to recommend NOP (in its existing state) for use on even a lightly loaded DVB-RCS network. The influence of traffic elsewhere in the network, and the inherent contended nature of the network, meant that performance would be poor and unpredictable. The MOS scores achieved were unacceptable in absolute terms, and are noticeably worse than the equivalent scores for “standard” VoIP traffic.

Crucially, the bandwidth saving gained by using NOP would be outweighed by the performance loss incurred.

7.2 Creation of Emulated Network

Whilst the Reference Network provided a real-life situation for testing, there was a need to carry out testing in an environment where different network conditions could be controlled and emulated. This facility was created by using a machine running the “Dummynet” software to reproduce the required (but now controllable) conditions.
The DVB-RCS link was replaced by a machine running the Dummynet software, which had two network adapters, and all IP traffic between the Remote Site and the Central Site passed through this machine. The Dummynet application therefore controlled the following network conditions (and others, if necessary), allowing their effects on the IP services to be monitored:

- Bandwidth.
- Packet loss.
- Latency (or delay).
- Jitter.

These effects could also be varied independently in both directions of traffic flow (to account for non-symmetrical link behaviour). The data collected in the characterisation task was used to program this Emulated Network.
The Dummynet software was not able to create jitter directly, so scripts were developed in order to simulate jitter. This was achieved by programmatically setting the latency to different levels while the test calls/procedures were carried out. The average difference between the latency levels is the simulated jitter.
The Emulated Network was used for all testing of the new software developments in the rest of the project.

7.3 Improve NOP Voice Quality

From the results of the testing, new algorithms were created for processing voice packets after they have been encoded from normal analogue speech, and before they are transmitted over the IP network. This made the voice traffic more resilient to the greater packet loss, delay, and jitter that are typically found on these links.
The following specific improvements were devised to make NOP a low-bandwidth alternative to standard RTP-based protocols:

1. A configurable jitter buffer was added to NOP. It has been seen that the use of jitter buffers helps non-NOP traffic. It was anticipated that this modification could help particularly at the margin of usable satellite links – in extreme low bandwidth cases, for example.
2. Side by side testing of NOP and H.323 showed that, all other factors being equal, H.323 can outperform NOP in terms of voice quality, *especially under difficult network conditions*. The only underlying differences between the two protocols are packet numbering and timestamping. Packet numbering (or “sequencing”) could be added to NOP without significantly adding to NOP’s bandwidth consumption. In order to support packet loss calculation and receiver reports, a packet counter (and timestamp) was added to the NOP packet frame structure.
3. Previously, the timestamps within RTP packets were “spoofed” by NOP before the packets were passed to other devices (external gateways, for example) for play-out. This method is not robust when delays become long and variable, as we have observed during the DVB-RCS testing – but adding internal timestamps would solve this problem.
4. It would be very valuable for the IPNx switch to be able to measure and report on attributes of the underlying transmission medium. Of course, the asymmetrical nature of DVB-RCS shows the importance of taking these measurements in both traffic directions. The measured characteristics would be available in logs for support engineers to consult in case of voice quality problems.
5. Sender and receiver reports proposed for NOP packets. By exchanging information on the quality of the reception, the NOP endpoints would be able to negotiate new parameters for the link dynamically in order to restore link quality. For example, NOP maximum packet size might be re-negotiated: lower packet size means there is less probability of a significant amount of voice being lost because of a random bit error; the end point would negotiate a new maximum packet size so that the packet loss is maintained below a certain level, while respecting an overall bandwidth requirement.
6. The variations observed in available bandwidth mean that packets are likely to be lost during a call because of buffer overflows. This makes the use of a voice codec that performs well under packet loss conditions highly desirable. The NetCoder proprietary codec claimed to offer superior performance in these conditions in comparison to the standard G-series codecs used in the tests carried out so far.

7.4 Improved Equipment Management

Current management tools have been extended to ensure that all operations can be carried out as effectively remotely as they can locally. The objective was that remote management should consume a minimum of the satellite link capacity, leaving as much bandwidth available as possible for revenue-earning voice calls.
The WTL equipment management protocol is called ‘wnmon’, and the following design parameters were used to make the necessary improvements:

- Low CPU Usage.
- Low Bandwidth Usage – no more than 5kbps or 3% of a link’s current bandwidth.
- Robustness – including a) respond rapidly to any requests that are made, and b) handle multiple simultaneous requests from different sources.
- Reliability – wnmon should run constantly, and must keep responding to requests even when the switch is filled to capacity.
- Resilience – wnmon should be as resilient as possible to malformed or potentially destructive requests.
- Compatibility with Other Protocols – wnmon must not impede any other IP-based protocol operations on a live IPNx switch.
- Easy Deployment in Secure IP Networks – wnmon should be available on a single IP port that can be easily forwarded or secured.

7.5 New Technique for Handling IVR (Interactive Voice Response)

The “Soft IVR” feature is an IP-based Interactive Voice Response (IVR) system for Pre-Paid customers that was developed to reside in the WTL switch at the remote site. The standard network configuration for Pre-Paid systems (before the development of Soft IVR) is shown in the following diagram:

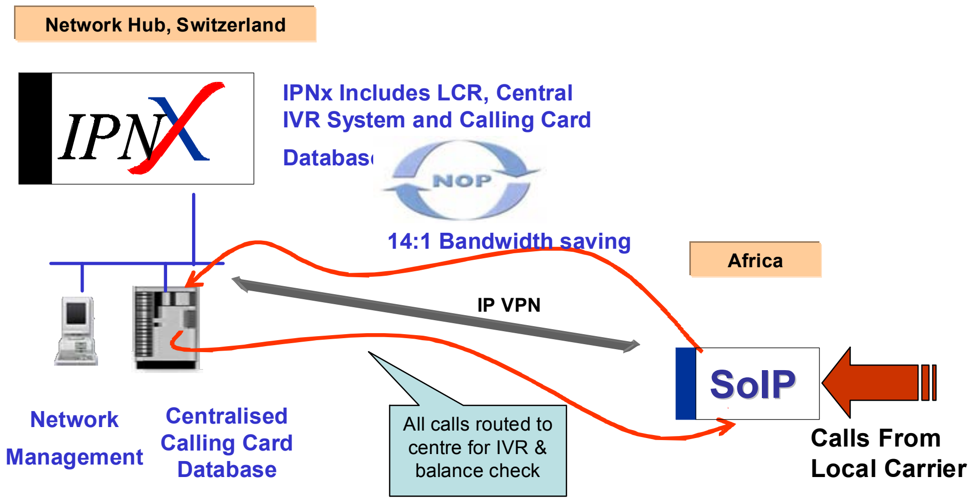

Figure 3 – Pre-Paid Network without Soft IVR

It can be seen that calls from the subscriber in Africa are carried all the way to the central account database for validation. The main reason for this approach is that the cost of authentication equipment in the remote location is high (dedicated DSP-based hardware is required). However, there are 2 major drawbacks:

a) The satellite bandwidth needed by the operator is increased (time on the link is occupied for the early stage of the call).

b) Centralising the caller authorisation function means more delay for the user, as message playback and digit recognition must come from the central site.

Research shows that as many as 30% of call attempts fail the authorisation step and do not lead to chargeable calls, so the above drawbacks are made even worse for the operator.

The WTL solution to this issue involves: a) IP message streaming; b) software-based DTMF digit detection. These are software-only functions, and can therefore reside in lower cost equipment at the remote end of the network, as shown in Figure 4 below:

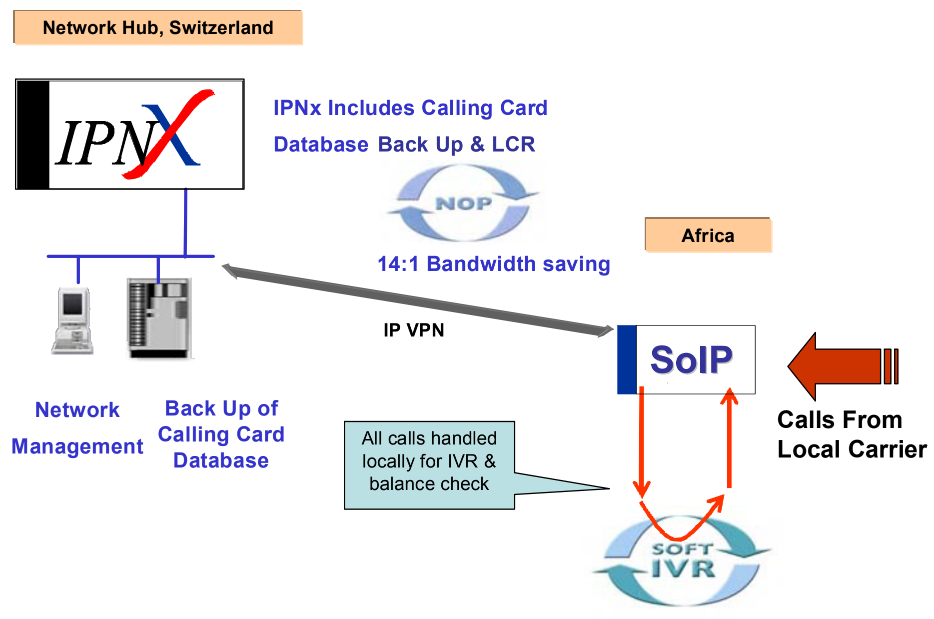

Figure 4 – Pre-Paid Network with Soft IVR

WTL already offer sophisticated database synchronisation and back up techniques, which will allow the remote database to operate as part of a larger, distributed, Pre-Paid network. The Soft IVR development will greatly reduce the cost and complexity of deployment, and also the monthly operating costs for companies offering Pre-Paid services.

7.6 Support for New High Quality/Low Bit Rate Codec

A major determinant of the voice quality delivered by a VoIP system is the voice encoding algorithm (codec) used. There are 2 common, standardised algorithms used at present by WTL and other VoIP equipment vendors: G.729 and G.723.1. These codecs attempt a compromise between voice quality and the amount of bandwidth used in their own ways.
As part of this project, WTL identified an alternative codec that is available from an existing technology partner: the 'NetCoder' codec from Audiocodes. According to an Audiocodes document, NetCoder is "a proprietary high quality, low delay and low complexity speech coder [that] is designed to provide the highest quality voice in Voice over IP network environments". The test results contained in that document show how favourably NetCoder performs compared to e.g. G.723.1 and G.729 in IP networks with high levels of packet loss.
NetCoder can be configured to use different amounts of bandwidth (6.4, 7.2, 8.0 and 9.6kbps), so that the operator can achieve the best compromise between bandwidth usage and voice quality. Such an approach fits well with the challenges presented by DVB-RCS networks that we were addressing in this project.
It should be noted that this task is quite independent of the improvements to NOP. The codec is responsible for the analogue to digital conversion of the voice samples, and as such is the baseline for the voice quality that the equipment can deliver (the voice quality can never be better than the chosen codec achieves). The maximum voice quality achieved by the codec is independent of the transmission technology used to deliver the samples.
NOP, on the other hand, is not concerned with the type of codec used, but deals only with the mechanics of transporting the samples.
Support for the NetCoder codec was added to the IPNx. Early tests showed that it did indeed give better MOS scores for a given bandwidth capacity compared to G.723.1 and G.729. Unfortunately, during Alpha testing, we learned that NetCoder will no longer be supported in future software releases from AudioCodes. At this point, therefore, it was decided not to include NetCoder support in the WTL products.

7.7 Independent Lab Tests for Voice Quality

It is difficult to measure voice quality objectively, so the decision was made to use a respected independent voice testing laboratory to verify the improvements made to the WTL products. Head Acoustics in Aachen, Germany (http://www.head-acoustics.de/) were selected for this testing. The choice was largely based on Head Acoustic's involvement with various ETSI-sponsored PlugTests, particularly the VoIP Speech Quality Test Events (SQTE).
A satellite IP network was modelled at the test laboratory as shown below, using simulation tools, based on the jitter, delay and packet loss parameter values measured by WTL earlier in the project.

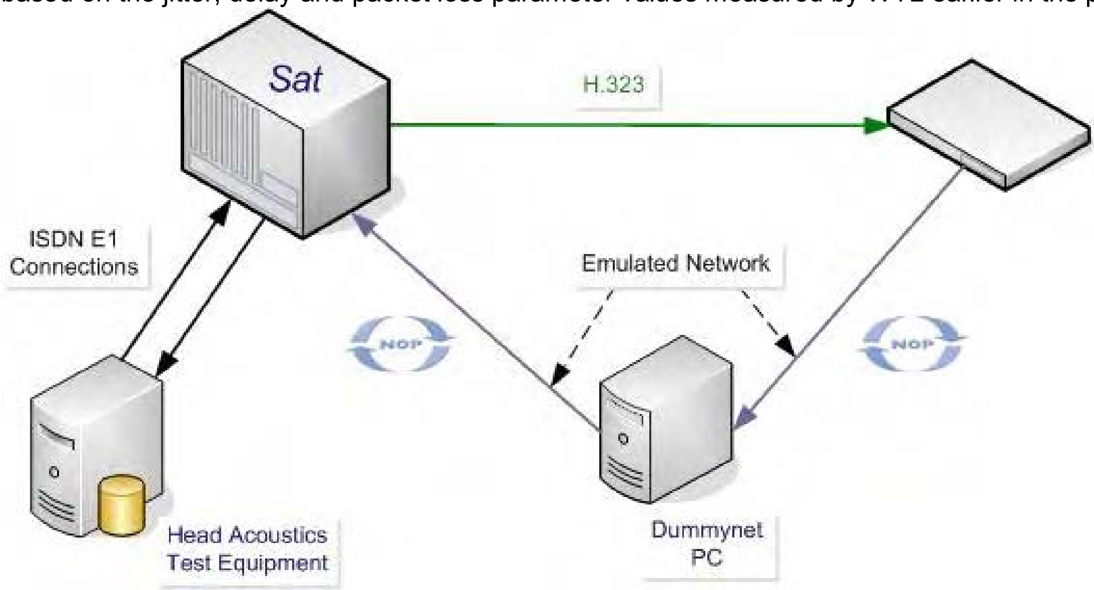

Figure 5 – Voice Quality Lab Testing Network

Measurements were taken for a series of representative usage cases. Automated MOS testing was used to show voice quality, and the bandwidth consumed on the IP link was measured for each case. The results are shown in a later section, but proved that a significant voice quality improvement was achieved as a result of this project.

7.8 Real World Customer Beta Test

One existing WTL customer was selected for the Beta Test. The purpose of the test was to confirm that the IPNx performed well in a live environment and that the package of enhancements in this project succeeded in creating a commercially viable product. The customer selected for the test was a leading wholesale carrier from Africa to Europe who used a VSAT (KU-Band, SCPC/DVB) link between Paris in France and Banjul in Gambia, via the NSS7 satellite.

In the Paris location, a WTL PVx (IPNx switch without compression hardware and/or TDM interfaces) had been installed that is connected to 3rd party equipment. The PVx's function here is to convert between the NOP and H.323 protocols in order to save bandwidth on the satellite link. The customer also uses the PVx as a calling card platform, so we were able to test the Soft IVR system there.

In the Banjul location, there is a SoIP Gateway, which is a smaller, PCI-X-based version of the IPNx. However, these products share identical software and a very similar physical architecture. The TDM interfaces of the SoIP Gateway were connected to several mobile phone operators. The call traffic flowed in both directions across the link, as shown in the following network schematic:

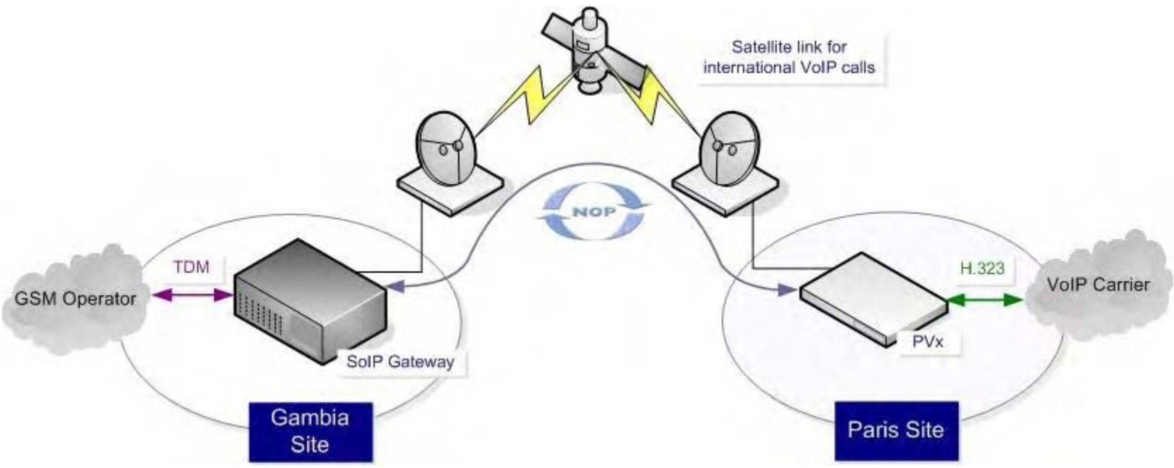

Figure 6 – Diagram of Beta Test Customer's Network

The Beta Test revealed a number of software problems and suggested enhancements, and these were included in the product before final release. At the end of the test the customer was satisfied enough to accept the modified equipment for live, commercial, operation.

8. Results

8.1 Voice Quality

The key result was proof that the **voice quality** of calls which involved the new NOP enhancements was acceptable for commercial use, and can save operators half the cost of their satellite bandwidth for carrying VoIP calls. This benefit holds true even in poor network conditions, as shown by the following tables.

Table 2 – MOS Scores for Original NOP vs. SIP/H.323

	Unloaded Network			Loaded Network		
Codec	**NOP**	**SIP/H.323**	**Diff (%)**	**NOP**	**SIP/H.323**	**Diff (%)**
G.723.1	3.1	3.6	-15.9	2.1	3.0	-40.2
G.729	3.0	3.1	-3.6	2.1	2.8	-38.0
G.711	3.1	3.5	-13.7	2.3	2.9	-23.3

Table 3 – MOS Scores for Enhanced NOP vs. SIP/H.323

	Unloaded Network			Loaded Network		
Codec	**NOP**	**SIP/H.323**	**Diff (%)**	**NOP**	**SIP/H.323**	**Diff (%)**
G.723.1	3.8	3.8	0.0	2.6	2.5	+4.0
G.729	3.8	3.8	0.0	2.8	2.9	-3.0
G.711	3.5	3.5	0.0	2.8	2.9	-3.0

A comparison of the tables shows that the enhancements made to the NOP protocol make it now perform at least as well as non-optimised VoIP calls. This means that the enormous bandwidth savings of NOP are delivered at little or no cost to the audio quality.

8.2 Bandwidth Utilisation

The following graph (Figure 6) uses data from the live customer Beta Test. It shows 18 simultaneous calls to/from Gambia using a mix of G.723.1 and G.729 traffic. It can clearly be seen that the bandwidth usage of NOP is half that of H.323 (around 200Kbps compared to over 500K).

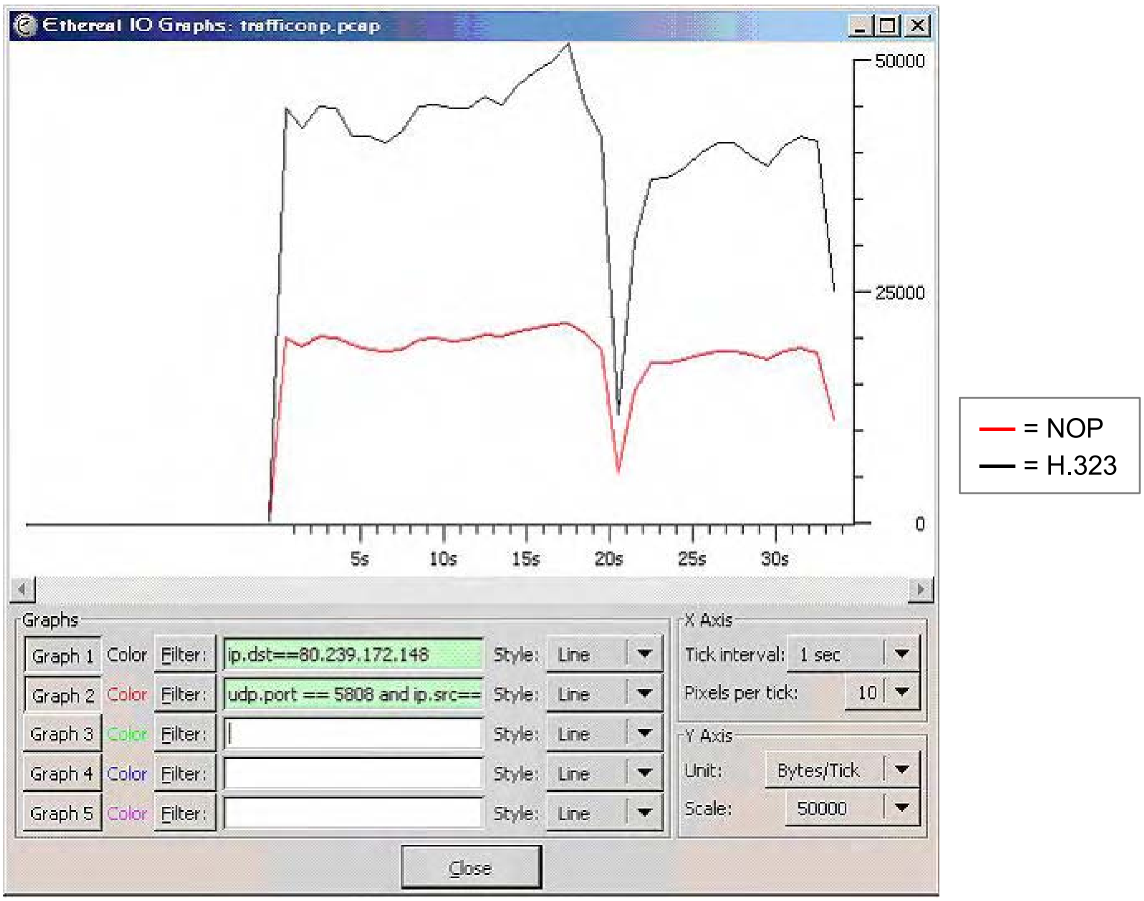

Figure 6 – Bandwidth Usage NOP vs. SIP/H.323

This equates to roughly 10Kbps being used per NOP call. This is not the lowest possible bandwidth consumption as this operator has chosen, for their own commercial reasons, to use both G.723.1 and G.729 codecs and not to enable silence suppression (this means packets are sent even when there is no speech).

8.3 Remote Management

Remote management of WTL equipment consumes less than 500bps in passive mode, but can burst up to 10Kbps when major updates or status enquiries are active.
The customer was highly satisfied with the **remote management** capabilities of the WTL equipment, and it was shown that there is no distinction between managing local or remote equipment. The performance of the remote management also held up well on heavily loaded or poor quality links.

8.4 Soft IVR

The **Soft IVR** system proved totally reliable, and should give major cost-efficiency benefits. Message and tone playing were consistent and of acceptable audio quality. DTMF digits were collected correctly before and during calls. Call start and end times were detected accurately so that call billing and call records were correct.

9. Financial Benefits

Taking a simple model the financial benefits of a WTL-based VoIP solution can be seen:

Table 4 - Factors for business case calculations

Basic Factors	Calc	Result
256Kbps DVB-RCS satellite link Monthly Cost	€4000	€4000
Bandwidth per NOP call	7Kbps	7Kbps
Max number of NOP calls on link	256/7	36
Typical billable minutes per call per month	10,000	10,000
Total billable minutes per month for this example (No. of calls * billable mins per call)	36*10,000	360,000

Prices charged and margins achieved will vary with time and according to local conditions, but the following is typical of an international Wholesale Operator:

Table 5 - Wholesale Business Case Calculation

Wholesale Financial Model	Calc	Result
Typical wholesale rate per minute to Africa	€0.08	€0.08
Typical wholesale margin	40%	40%
Typical wholesale profit per minute	40% * €0.08	€0.032
Wholesale margin earned per month (Billable Minutes * Wholesale Profit)	360,000 * 0.032	€11,520
Subtract cost of satellite link	- €4000	- €4000
Wholesale Net Profit	€11,520- €4000	**€7,520**

In a Retail service, other costs like promotion, distribution, and customer service must be considered, but the following is a simplified example of a Retail operation:

Table 6 - Retail Business Case Calculation

Retail Financial Model	Calc	Result
Typical retail rate per minute to Africa	€0.16	€0.16
Typical retail margin	50%	50%
Typical retail profit per minute	50% * €0.16	€0.08
Retail margin earned per month (Billable Minutes * Retail Profit)	360,000 * 0.08	€28,800
Subtract cost of satellite link	- €4000	- €4000
Retail Net Profit	€28,800- €4000	**€24,800**

Both business cases generate profit, but a key reason for this is the use of low cost DVB-RCS services.
The effect of using NOP is clear in both cases: by doubling the number of calls that can be carried, the maximum achievable revenue is correspondingly increased. By utilising Soft IVR, the Wholesale operator has the opportunity to offer higher margin Pre-Paid services on the same piece of equipment at no extra hardware cost.

10. Conclusions

This project has shown that there is a very strong future for VoIP over DVB-RCS satellite services, and the WTL equipment is now very well adapted to this purpose.
The ability to use optimised VoIP over DVB-RCS gives long distance telecom operators many new opportunities. Wholesale operators can make major savings carrying bulk traffic in and out of destinations in Africa and Asia. The Retail segment of the telecom market will enjoy the bandwidth savings, but also the use of Soft IVR to deliver a low cost Pre-Paid service locally is a breakthrough in cost terms. The lower cost of infrastructure and space segment is likely to be particularly appealing to less developed countries where telecom provision is still sparse, and significantly lowers the barriers to entry for new operators.

The benefits of VoIP over satellite – and WTL equipment in particular – can be summarised as follows:

- Good voice quality (MOS rating up to 3.8).
- Voice quality consistent over time.
- Usable voice quality even in poor line/network conditions.
- Independent testing proves the voice quality claims.
- NOP allows low satellite bandwidth consumption (less than 7Kbps per call in best cases).
- WTL equipment offers carrier-grade reliability.
- Powerful, user-friendly equipment management is possible using little bandwidth.
- Management expertise does not need to be at remote site.
- Soft IVR gives all-in-one solution for Pre-Paid, VoIP switching, routing, billing, and traffic optimisation.
- Pre-Paid services can be offered with low cost equipment, and therefore at a lower risk for operators.

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