Certificate of Compatibility

NEC Infrontia Limited is pleased to verify that:

Xeloq Communications (Netherlands)

has successfully met the standards for SIP Trunk compatibility
with the NEC Infrontia products listed below.

- Aspire
- XN120

Test Completion Date:	24^{th} May 2006
Test Location:	Remote
Name of Provider:	Xeloq Communications
Website:	http://www.xeloq.com
System Tested:	Aspire
Software Version:	5.22
SIP Connection Mode:	Carrier
Test Plan Version:	1.1

Please refer to the following page(s) for further
information and Configuration Notes.

Disclaimer:
NEC Infrontia Limited has performed Interoperability Testing with the Provider listed above. The results of these tests proved satisfactory.

IMPORTANT: NEC Infrontia Limited cannot be held responsible for any future compatibility issues that may arise, as Providers may make changes to their systems which are outside of NEC Infrontia's control.

SIP Configuration Notes – Xeloq Communications

This configuration guide should be used to configure an Aspire/XN120 system for connection to the service described above, via SIP trunks.

Recommended Software Versions

Aspire:		*XN120:*	
NTCPU	V5.24	CPU	V5.23
PCPro	V6.10	PCPro	V5.01
VOIPU	V4.05	VOIPU	V4.05

System Configuration

The following items should be changed – all other items are considered irrelevant or should be left as default.

PRG	Easy Edit	Item	Setting
10-12-01	2955	IP Address	Set according to customers network requirements
10-12-02	2955	Subnet Mask	Set according to customers network requirements
10-12-03	2955	Default Gateway	Set according to customers network requirements
10-03-02	1065	Trunk Type	Set VOIPU card trunk ports to SIP
10-28-01	1040	Domain Name	Set according to customers network
10-28-02	1040	Host Name	Set according to customers network
10-28-04	1040	UserID	The account number assigned by Xeloq
10-28-05	1040	Domain Assignment	Set to IP Address
10-29-01	1041	Default Proxy tx	On
10-29-02	1041	Default Proxy rx	On
10-29-05	1041	Register Mode	Manual
10-29-08	1041	DNS Server Mode	On
10-29-09	1041	DNS Server IP	Customers DNS Server IP address
10-29-11	1041	Registrar	Set to sip.goandcall.com
10-29-12	1041	Domain Name	Set to goandcall.com
10-29-13	1041	Host Name	Set to sip
10-29-14	1041	Carrier Choice	Carrier Choice 1
10-30-02	1042	Username	The account number assigned by Xeloq
10-30-03	1042	Password	The password assigned by Xeloq
If multiple accounts have been provided the following 4 items should be configured for each account			
10-36-01	1070	Registration	Enable
10-36-02	1070	User ID	The account number assigned by Xeloq
10-36-03	1070	Auth User ID	The account number assigned by Xeloq
10-36-04	1070	Auth Password	The password assigned by Xeloq
84-05-01	1052	IP Address	Set according to customers network requirements
84-13-28	1057	Audio Capability	Set codec according to customer requirements
84-13-32	1057	DTMF Relay Mode	Set to RFC2833

- DDIs can be configured if required, using the same procedure as for ISDN trunks
- SIP calls are sent “en bloc”. This means that the External Call Interdigit timer (PRG21-01-03, Easy Edit page 1474) must expire before the call is set up. This can be reduced, but will have an impact on ISDN trunks also. The user can dial # to indicate “end of dialling” instead if required.

Network Configuration

If Public IP addresses are assigned to the Aspire NTCPU and VOIPU, then there should be no network configuration required.

If there is one public IP address assigned, and NAT is used, it is necessary to configure Port Forwarding on the router:

- Port 5060 should be forwarded to the NTCPU IP address
- Port 10010 – 10100 should be forwarded to the VOIPU IP address
- NAPT should be enabled in PRG10-12-06 and the Public IP address should be entered into PRG10-12-07. (Easy Edit page 2955)

Known Limitations

- If calling via the PSTN, it is not possible to withhold CLI. The call will be rejected if the caller ID is being withheld on the telephone system
- There is a direct DDI to Account number mapping. This means that if multiple DDIs are required, then multiple accounts need to be registered. This limits the number of DDIs to 32 (the maximum number of registrations in the telephone system)
- The telephone system can manipulate the CLIP for outbound calls. However, this is not passed through to the called party. This is a limitation of the SIP carrier.

  Eg.

International calls:	No CLIP is sent
National calls (PSTN):	00<AccountNumber> is sent (eg. 00706000)
SIP Calls:	<AccountNumber> is sent (eg. 706000)

- G.711 is not supported

Comments

- Tones are sent within RTP, rather than via signalling. For example, when making an outgoing call the Ringback Tone is sent by the SIP server as RTP. This means that the tones are susceptible to poor network conditions and that VOIPU resources are used even during the signalling phase. This should not cause a major problem in most circumstances

Document History

Version	Date	Description
1.0	12th June 2006	Initial Release