Alcatel·Lucent

OmniPCX Enterprise

USING Audiocodes MP11x FAX G711 TRANSPARENT MODE WITH OmniPCX Enterprise WITH SIP Carrier SIP Trunks : OBS BIVSIP S2

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☐ URGENT

■ NOT URGENT ☐ TEMPORARY ■ PERMANENT

SUBJECT: USING FAX G.711 TRANSPARENT MODE WITH OmniPCX Enterprise WITH CARRIER SIP TRUNKS

CONTENTS

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1. INTRODUCTION

The aim of this document is to provide information relative to the configuration of Alcatel-Lucent OXE R11.1 IPBX embedded in OTBE R2.1 and dependent AudioCodes MP-11x fax gateway as G.711 passthrough mode (aka G.711 transparent fax) with Orange Business Services BIV SIP S2 Network Offer.

This configuration is valid for OBS eDiatonis IP MultiSites R2.1 (aka IPMS R2.1with embedded OXE R11.1) offer and OTBE R2.1 generic offer within the same features parameters as IPMS R2.1.

OmniPCX Enterprise (OXE) system supports 2 FAX Over IP modes i.e T.38 protocol (ITU-T T.38 recommendation (From R7.1) & G.711 transparent fax (from R11.0) for SIP carrier's connections.

In case of OBS BIV SIP S2 connection, as only Transparent Fax G.711is supported, G.711 transparent fax configuration is mandatory.

In this context, two scenarios for connecting analog faxes are available:

Note: OTBE embedded fax server is not validated on BIV SIP S2 connection.

- Analog Fax behind OXE IP MG with G.711transparent fax configured on OXE SIP trunking connection. (Not part of this document, refer to OXE technical documentation)
- Analog Fax behind a third party ATA Fax gateway box (e.g. Audiocodes MP11x) configured to interwork with SIP trunking BIV SIP connection in transparent fax G.711 and analog Faxes declared on OXE as SIP devices.

This document details this last solution design, the related constraints, restrictions, and its implementation in OXE management (first configuration part of this document) and Audiocodes MP11X configuration (second configuration part of this document).

OXE management part intends to bring help to the OXE administrators to understand and implement necessary management rules on OXE side in order to force G.711 use for calls initiated from or to Third party ATA Fax gateway box. Once G.711 RTP flow is setup, Fax call completion and quality in G.711 transparent mode is dependent on carrier network characteristics.

Third party ATA Fax gateway box part concerns the Audiocodes MP-11x configurations set up and validated by OBS BIV SIP labs during IPMS R2.1 homologation process.

Note that this part:

- Shows the Audiocodes configuration through graphical user interface only. It is possible to perform the configuration of Audiocodes MP-11x gateway through command line interface, although this is not described in this guideline.
- Does not concern Analog phones as it has not been part of OBS BIV SIP validation.

2. REFERENCES

[1] Omni PCX Enterprise R11.x Documentation

[2] Audiocodes Alcatel-Lucent Application Partner Programm (AAPP) MP112 Inter Working Report (OXE R11 MP11x version 6.6)

3. OXE CONFIGURATION DESCRIPTION

3.1 Global overview of the solution

The solution described in this document is based on the combination of the Omni PCX Enterprise (Embedded or not in an OTBE) and third party ATA Fax gateway box products. Fax gateway is in charge to connect legacy Fax devices to the OXE system using SIP protocol.

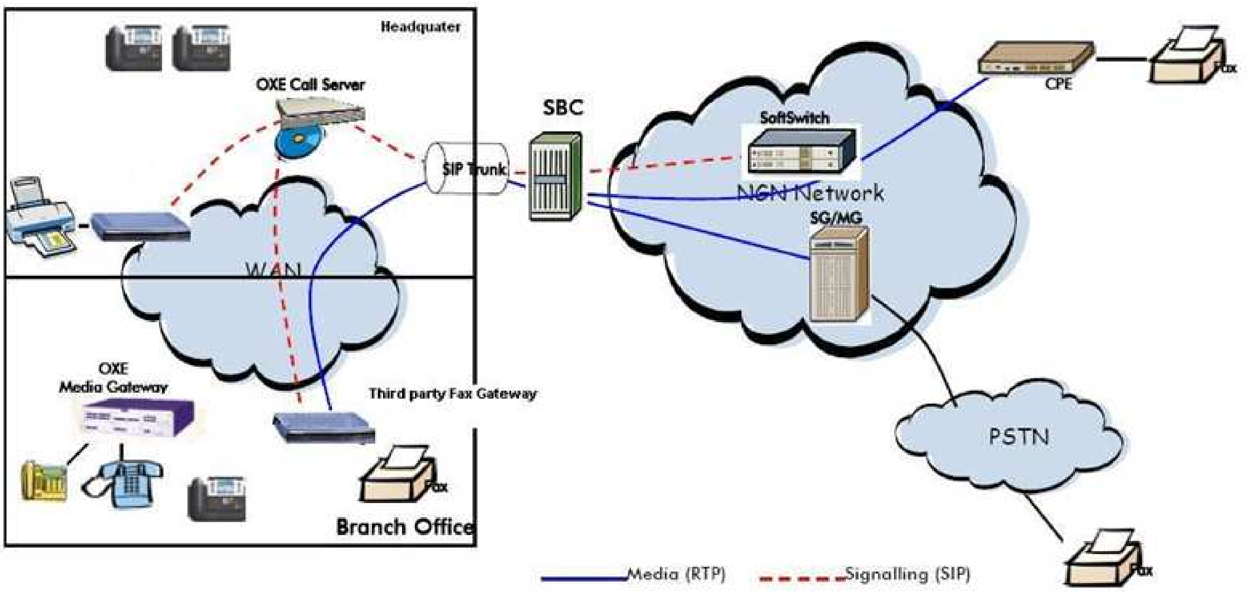

Figure 1: Global solution scheme

3.2 Solution perimeter

This document describes solution design for OmniPCX Enterprise to offer Fax G.711 feature for inter-working with NGN network through SIP trunking.

This document describes the supported scenario and the specific configuration and limitations associated.

As a general rule, OXE system configuration must comply with the specifications defined by document [1], except when explicit specific configuration is mentioned in the present document. Fax gateway configuration must comply with the documentation provided by Third Party Company. This configuration is out of the scope of Alcatel-Lucent configuration part as parameters should be adapted to carrier network characteristics. Alcatel-Lucent only provides OXE engineering rules to be deployed in order to ensure an adequate SIP session establishment between Fax gateways and carrier border element, suitable for Fax G.711 operation.

Third Party Company is responsible for providing information related to its product, regarding Voice and Fax transparent mode for configuration, restrictions and limitations.

3.3 Key factor for successful G.711 feature inter-working

Fax G.711 inter-working feature needs the following requirements to be strongly realized in order to ensure a successful operation:

- Media stream (RTP stream) must be directly established between endpoints (at least between Third party ATA Fax gateway box and carrier SIP border element through SIP trunk), when the Fax signal starts. In other words, no intermediate OXE IP Media Gateway must be used when Fax communication is involved. To ensure the RTP path will be direct, some constraints have to be enforced regarding session establishment (SIP protocol) and domain management
- Communication from and to the Fax gateway must starts using G.711 audio from the beginning of the call. Adequate configuration will ensure communication starting in G.711 from the beginning.

3.4 The expected RTP flows

In a global corporate solution, the OXE system configuration for G.711 transparent fax enablement shall lead to support the following RTP flow:

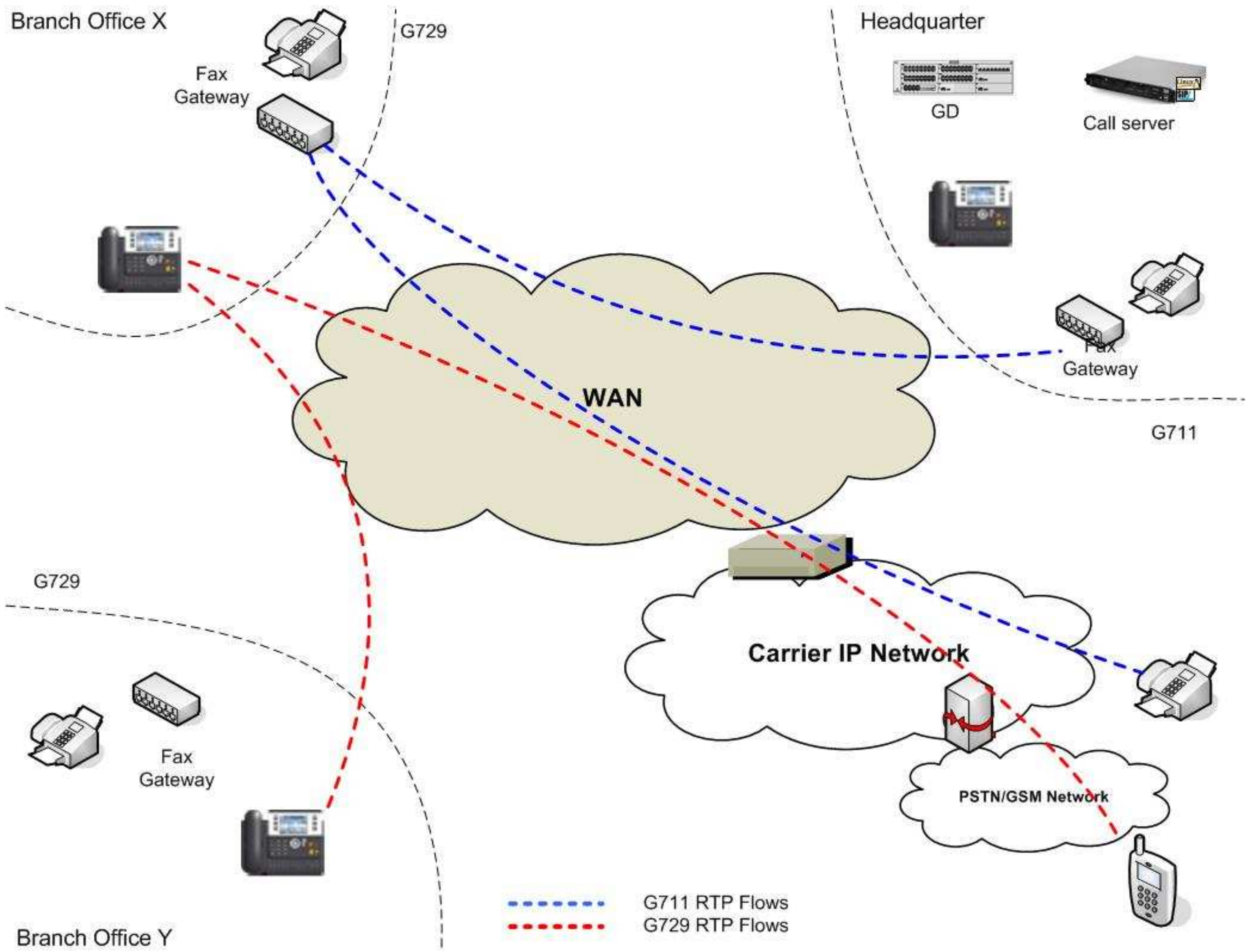

Figure 2 Expected RTP flows for fax G.711 support

Headquarter has non restricted access to WAN and can use G.711 for voice calls to other locations or external numbers. Here voice and fax calls will use same G.711 codec, if remote party supports it for voice calls.

For branch offices at location X and Y, less bandwidth is available and G729 codec is used for voice calls to other locations. For fax calls, system management must ensure that G.711 will be used between fax gateways and remote party.

3.5 Media negotiation on the public SIP trunk

FAX gateway management and OXE management must ensure that G.711 will be selected at call establishment (during media/codec negotiation phase), for both directions:

– Fax sent from a fax behind a fax gateway to an external device in carrier network
– Fax received on a fax behind a fax gateway from an external device in carrier network

On the opposite, any voice call from a restricted domain should still use G729 if carrier proposes it.

In this example, carrier supports both G729 and G.711 codecs like OBS BIV SIP S2 infrastructure.

3.5.1 Media negotiation rules / Overview

By default on public SIP trunk, OXE try to maximize inter-working success by systematically offering a wide list of supported codecs.

The list of codecs, which is proposed for outbound calls depends on the relevant gateway parameter ("**Type of codec negotiation**"), the system compression codec ("**Compression type**" / G729 or G723), and the “**multi algorithm**” parameter.

We suppose here that the "**Compression type**" parameter is managed to G729, and that the “**multi algorithm**” parameter is set to false. Then, depending on the management:

- Type of codec negotiation : Default
  Proposed list of codec on SIP trunk : G.711 , G729

- Type of codec negotiation : G.711 only
  Proposed list of codec on SIP trunk : G.711

- Type of codec negotiation : G729 only
  Proposed list of codec on SIP trunk : G729

When IP domains are managed, the outbound calls are by default considered as extra domain. Therefore, the original codec list described above might be re-ordered properly, e.g. the extra domain codec might be inserted first, depending on the IP domain the caller belongs to.

3.5.2 Media negotiation for fax G.711

The FAX Gateway has to be declared as SIP Device.

As communication start up with G.711 is a key factor for fax support, system engineering rules must limit codecs list proposed during codec negotiation on public SIP trunk side.

Regarding the SIP carrier on an OXE management point of view, two separate External Gateway are used:

- Incoming gateway to handle voice outgoing calls and all incoming calls
- Outgoing gateway to handle FAX outgoing calls

Both gateways share the same management regarding the “**Remote Domain**” and “**Outbound proxy**” (If any) parameters.

This section focuses on this outgoing gateway.

For incoming calls (SIP trunk to FAX Gateway), this must be done in SIP device configuration, limiting codec list to G.711 only, in OXE system law (A/mu).

For FAX outgoing calls, the "***Type of codec negotiation***" dedicated parameter of the outgoing gateway on OXE has to be managed to “**G.711 only**”, in order to restrict the OXE to provide a single G.711 (A or µ) offer during SIP session negotiation between OXE system and carrier SIP network access element.

For Fax G.711 feature, this parameter is used to ensure direct RTP and G.711 communication from the beginning. To summarize, in order to be able to use G.711 only in both directions (outgoing/incoming calls) configuration requirements are the following:

- The "***Type of codec negotiation***" parameter of the External gateway used for outgoing fax calls **MUST** be configured as "***G.711 only***"
- The "***Outbound calls only***" parameter of the External gateway used for outgoing fax calls **MUST** be configured as "***Yes***"
- The "***FAX procedure type***" parameter of the External gateway used for outgoing fax calls **SHOULD** be configured as "***T38 only***", although it remains not significant in such a configuration
- Fax gateway **MUST** be internally configured to offer/accept **only** G.711
- If IP domains are involved in the Fax G.711 , theses IP domains **MUST** allow G.711

Carrier SIP trunk **MUST** support and accept G.711 and **MUST** offer at least G.711 in its SIP/SDP codec lists.

Here is an example of an incoming call to Fax gateway, taking into account this requirements:

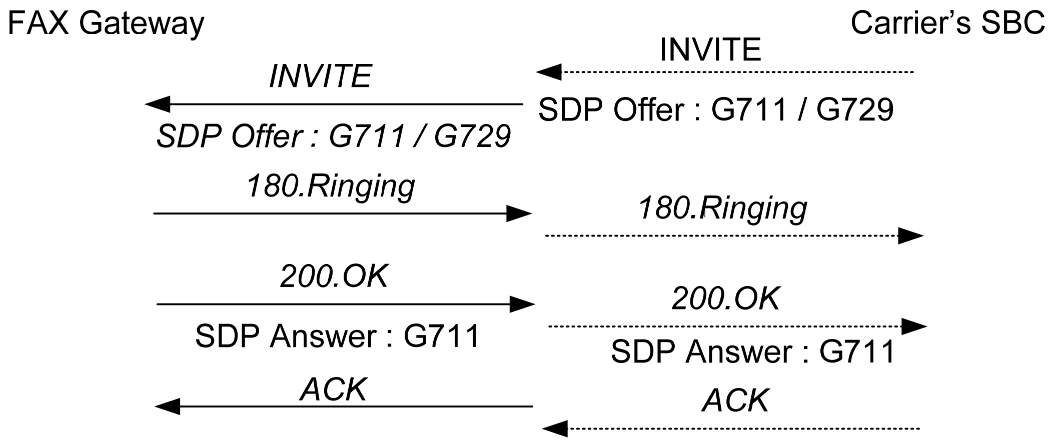

Figure 3: Incoming call

For outgoing communication, thanks to this configuration, OXE will only offer G.711.

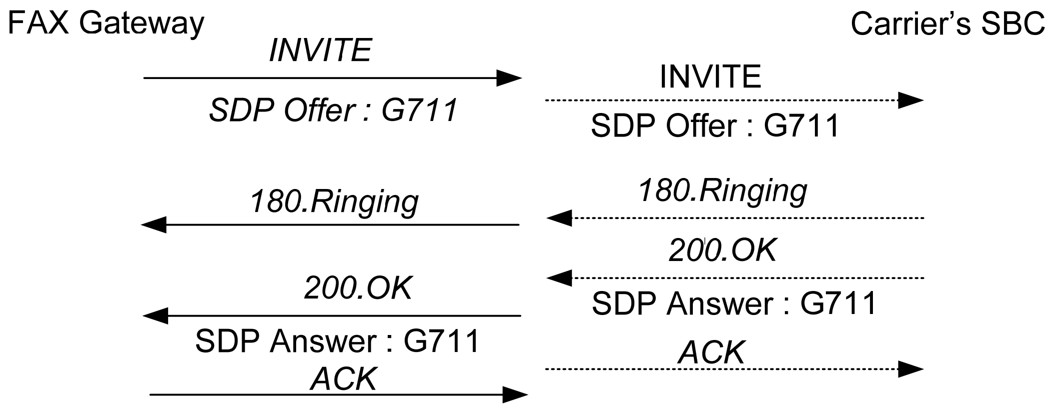

Figure 4 : Outgoing call with forced use of G.711 on public SIP trunk

3.5.3 Constraints on SIP/SDP session establishment

3.5.3.1 VBD, SilenceSupp and ecan SDP attributes

SDP VBD, SilenceSupp and ecan media attribute are not supported. The OXE SIP Proxy silently discards them.

3.5.3.2 Re-INVITE after session establishment

If subsequent SIP INVITE (RE-INVITE) messages are presented by carrier network session border controller, these messages will be transparently relayed to the Fax gateway by OXE SIP proxy in the following cases:

- A new codec is requested
- A new TSAP is requested (IP address or RTP port number)
- A new payload type for voice is requested
- A new payload type for telephony-event is requested

As a consequence, such re-INVITE should not be sent by carrier network in order to avoid any fax transmission failure.

3.6 OXE management rules

3.6.1 IP domains

Note: IP Domains configuration for Remote Sites Not Applicable in case of OBS BIV SIP connection as BIVSIP Offer is dedicated to mono site customer.

If strict bandwidth management is required (CAC), IP domains usage is mandatory. When Fax gateways and voice terminals with low bit-rate codec are installed in the same site, Fax gateways **MUST** be managed in a separated domain, dedicated to local Fax machines (group of Fax is allowed, but mixture of Fax and other equipment is forbidden).

Extra domain codec must be managed as "**without compression**" for fax domain. CAC is managed with CAC counter of each domain: fax domain, and voice domain.

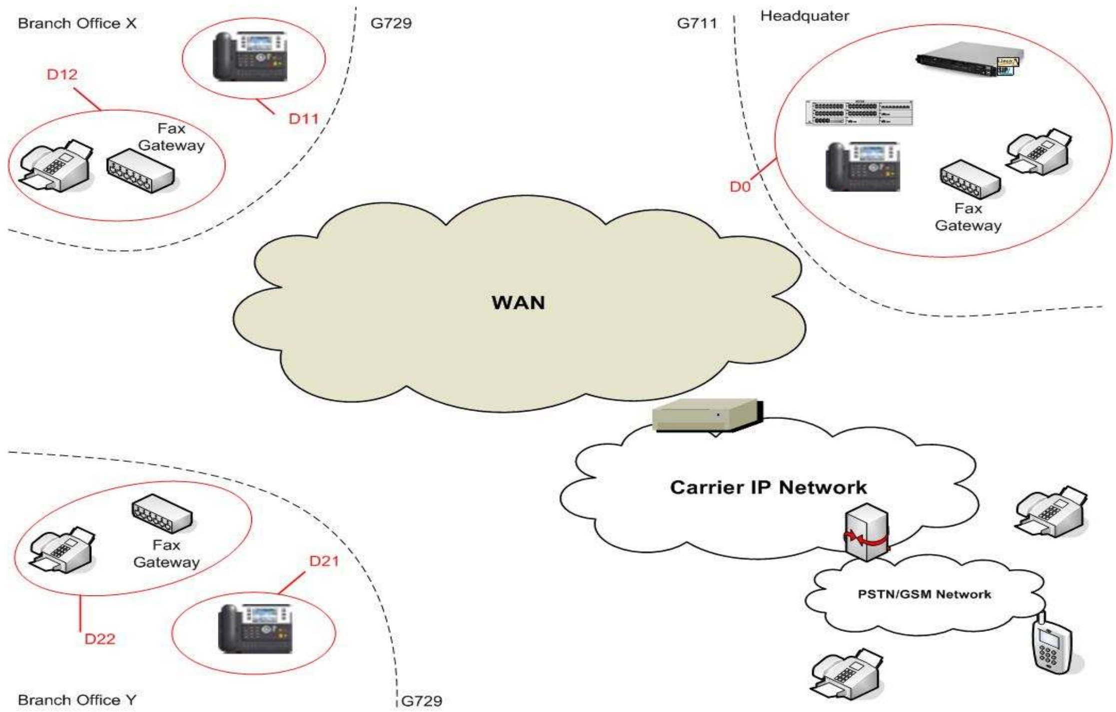

Figure 5: Domains management

In this example the following management applies:

- For headquarter:
  - Call servers must belong to default domain 0
  - Voice devices and Fax devices are also managed in domain 0, as G.711 can be offered with high bandwidth access to WAN. Domain 0 compression parameters are:
    - ♦ Intra domain: without compression
    - ♦ Extra domain: without compression
- For Branch office X:
  - Voice devices are managed in domain 11 with compression parameters:
    - ♦ Intra domain: without compression
    - ♦ Extra domain: with compression
  - Fax devices are managed in domain 12 with compression parameters:
    - ♦ Intra domain: without compression
    - ♦ Extra domain: without compression
- For Branch office Y:
  - Voice devices are managed in domain 21 with compression parameters:
    - ♦ Intra domain: without compression
    - ♦ Extra domain: with compression
  - Fax devices are managed in domain 22 with compression parameters:
    - ♦ Intra domain: without compression
    - ♦ Extra domain: without compression

Then, for branch office with restricted bandwidth, OXE will handle CAC for each kind of call separately, with the following consequences:

- Global CAC limit for the site is given by: CAC_limit = CAC_fax + CAC_voice where CAC_fax is the CAC counter for fax domain, and CAC_voice is the CAC counter for voice domain.
- Global bandwidth on WAN access point must be calculated, taking into account client needs for each kind of call.
- Voice calls between Fax gateways and voice terminals on the same site would decrease CAC counters for both domains and should be forbidden. The other case lead to the following consequences:
  - CAC counters couldn't be used to manage WAN access. Nevertheless, this will decrease number of calls on access, but never leads to CAC overload situations.
  - Transcoding resources would be allocated by OXE to set up such calls, leading to unexpected compressor allocation. These resources would be taken in a domain without compression for

extra domain calls. It could be domain 0 in our example, or any centralized domain dedicated to transcoding and tone/voice guide playing.

CAC management for sites including G.711 fax and low bit rate voice terminals has been improved. In addition, domains can be managed as "Tandem" domains, taking into account global site call limit on one hand, and fax/voice calls relative cost on bandwidth consumption on the other hand. The following management rules apply:

- Both domains must be managed with the same extra domain coding algorithm value: "with compression"
- Voice domain must be declared as Tandem primary domain, i.e "Tandem primary domain" parameter set to (-1)
- Fax domain must be declared as Tandem secondary domain, i.e "Tandem primary domain" parameter set to Tandem primary domain (i.e voice domain) number
- In voice domain, "Domain Max Voice Connection" must be set to represent global site bandwidth, if only voice calls are done.
- In Fax domain, "Domain Max Voice Connection" must be set to represent global site bandwidth, if only fax calls are done.
- In Fax domain management "Tandem CAC factor" must be set to represent fax calls relative cost compared to voice calls.

In our example we could manage domain 11 and 12 the following way:

	Domain 11 (voice)	Domain 12 (fax)
Domain Max Voice Connection	9	3
Tandem Primary domain	-1	11
Tandem CAC factor	-	3

In this example, if a voice call is done, voice domain CAC counter is incremented with 1. If a fax call is done fax domain CAC counter is incremented with 1 and voice CAC counter is incremented with 3. If one the counters exceeds the related domains limits, the call is rejected.

If a voice call is performed between a fax gateway and a voice terminal inside a couple of domains managed in "Tandem", this call is considered as intra-domain and CAC counters not incremented.

3.6.2 Use of a specific external gateway

As described in 3.5.2, in order to restrict codec list content to G.711 for outgoing calls, the "**Type of codec negotiation**" external gateway (To be used for FAX calls) parameter has to be set to "**G.711 only**". In order to be able to process voice calls with complete codec list on the same system, two different external gateways must be used for voice and fax communication.

In the above example, calls from fax gateways (in D12, D22, and part of D0) must use the "**G.711 only**" external gateway. On the opposite, calls from other voice extensions, should be set up with complete codec list, built and ordered depending on extra-domain constraint.

As outgoing external gateway is selected by ARS route CDT, two different ARS routes must be managed, on for each kind of extensions (Fax/voice). Fax gateways and other voice terminals should be managed in two different entities in order to use dedicated ARS routes and external gateways.

In our example, the following managements could be proposed:

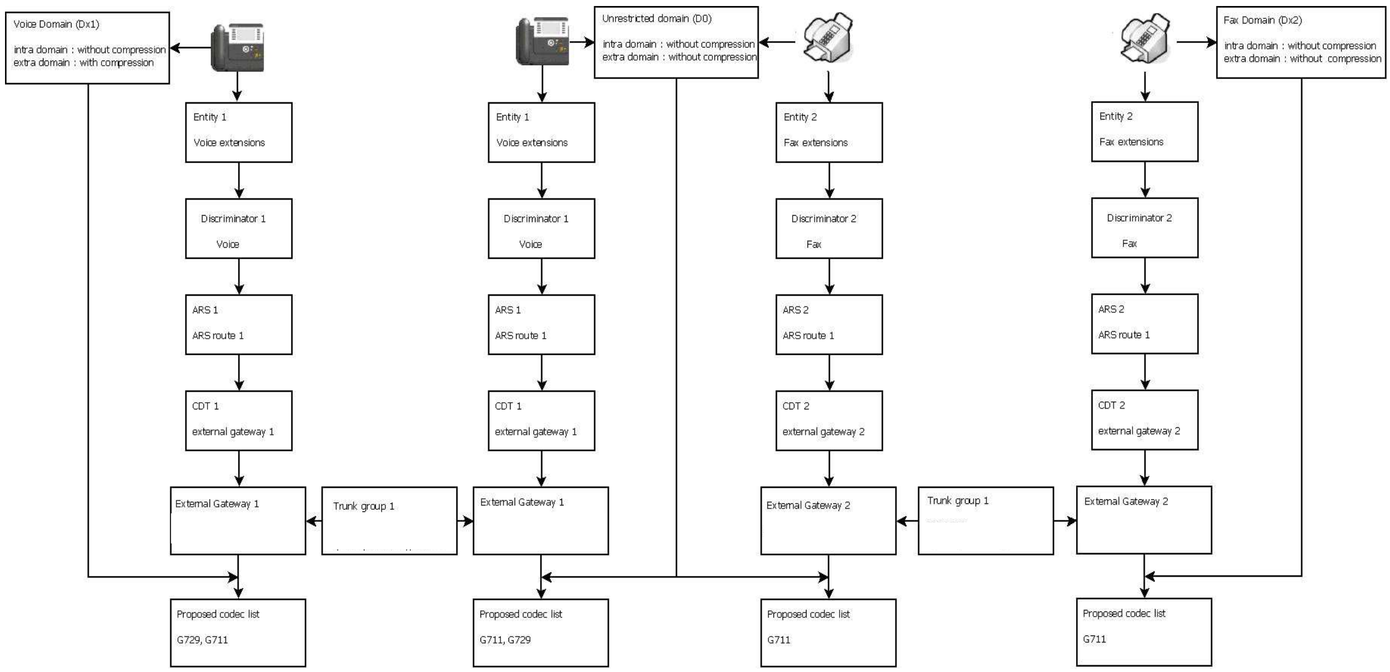

Figure 6: Management for fax codec list building on outgoing calls

3.7 Restrictions

3.7.1 Technical restrictions

Inter-working between Fax G.711 and other OXE Fax transport mechanism (T38, proprietary Fax transparent) is **not** supported.

3.7.2 G.711 fax Troubleshooting and support

In client site deployment scope, Fax gateways used for G.711 transparent fax should be part of Alcatel-Lucent Alliance and Application Partner Program (AAPP) in order to ensure basic inter-working with OXE using SIP protocol.

Fax G.711 with Third party ATA Fax gateway box feature is not part of AAPP program testing with OXE.

This feature is supported for specific offers with SIP carriers.

Fax gateway configuration is identified during specific test session in front of carrier network. An interoperability guide is then defined, indicating the configuration of the third party product after certification tests.

- Any issue in G.711 session establishment from/to the fax gateway, in an OXE configuration compliant with the present document, should lead Business Partner to request Alcatel Technical Support.
- Fax transmission completion and quality issues should be troubleshoot with the help of Application Partner responsible for fax gateway on one side, and carrier providing network access with SIP trunking on the other side.

Please note that Fax G.711 transparent is less resilient than T38 in case of bad network conditions (packet loss, jitter) and has no redundancy mechanisms to overcome these restrictions. Defining specifications for maximal network IP packet loss, delay, jitter is out of Alcatel Lucent responsibility.

4. AUDIOCODES MP11X FAX GATEWAY CONFIGURATION

This part describes the main configuration for G.711 fax Transparent on MP11X.
For generic configuration of MP11x see [2]

4.1 Network configuration

4.1.1 IP configuration

Connect to the administration web interface (http://AudioCodes_IP_address). If AudioCodes has never been configured, default factory IP address is 10.1.10.10. Log on the system ("Admin"/"Admin"). On left-frame, select the full menu.
Select "Configuration" tab in order to display the configuration menu. Expand "Network Settings" object and click on "IP Settings" menu. Declare IP address of MediaPack equipment, its associated subnet mask and the IP address of the default gateway.

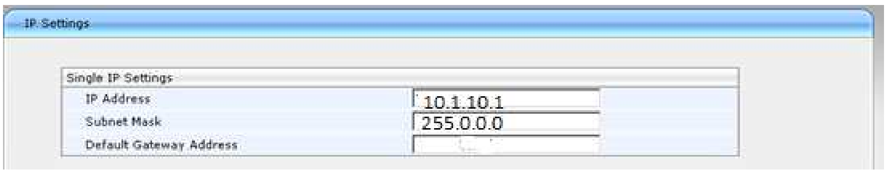

Figure 7: IP Configuration

4.2 SIP configuration

4.2.1 Protocol configuration

Select "Configuration" tab. Expand "Protocol Configuration" and "Protocol Definition" and click on "SIP General Parameters" item. "Early media" must be enabled. 183 messages consist in "progress" messages. Session expiration method consists in sending an UPDATE. Fax signaling method must be configured as "G.711 transport". Fax must be detected on preamble. SIP local and destination ports must be configured with their default values (5060 for TCP/UDP and 5061 for TLS). Tel URI must not be mentioned in asserted identity header.

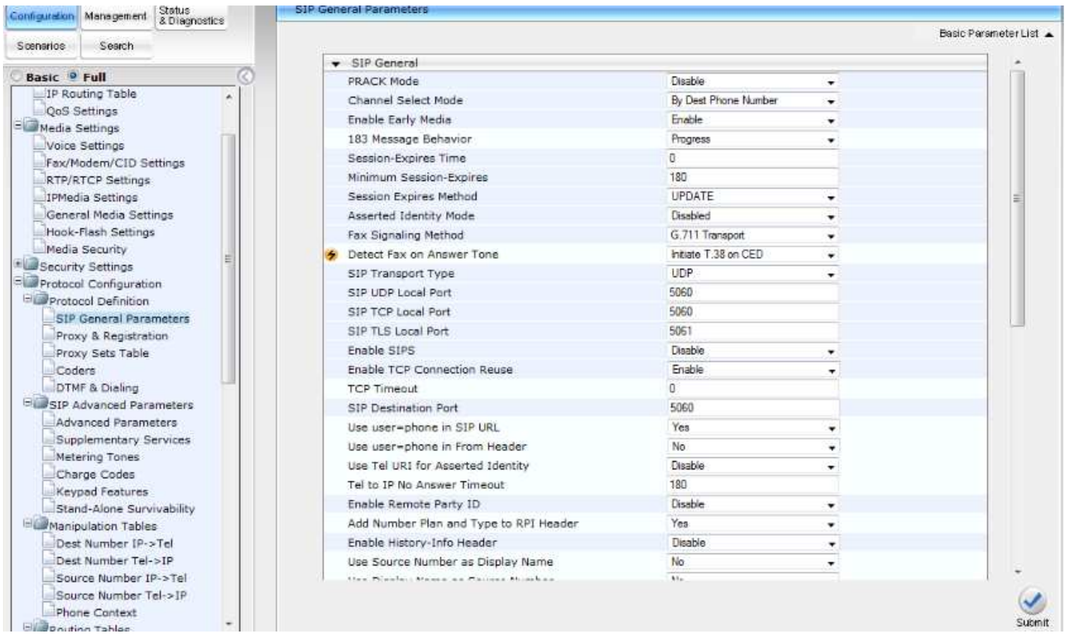

Figure 8: SIP configuration (1/2)

History-info field must be disabled on MediaPack gateway. RBT mustn't be played to IP. RBT to fax device will be played by MediaPack gateway: "Play According to Early Media".

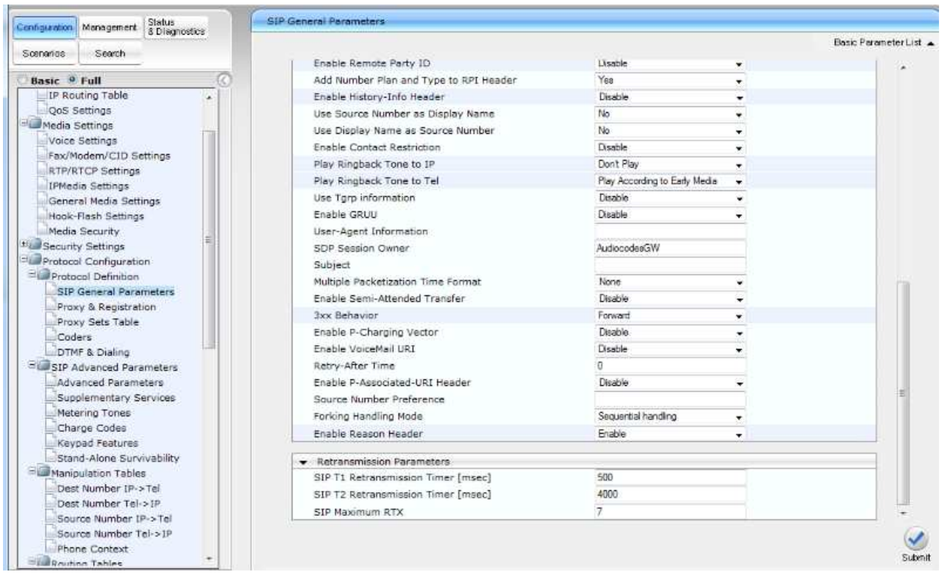

Figure 9: SIP configuration (2/2)

Select "Configuration" tab. Expand "Profile Definitions" objects and click on "IP Profile Settings" item. Echo canceller is "Enable" when G.711 transport mode is configured. Fax Signaling Method is "G.711 Transport". Play Ringback Tone to IP is "Don't Play". Enable Early Media is "Enable"
Dynamic Jitter Buffer Minimum Delay is "70" ms

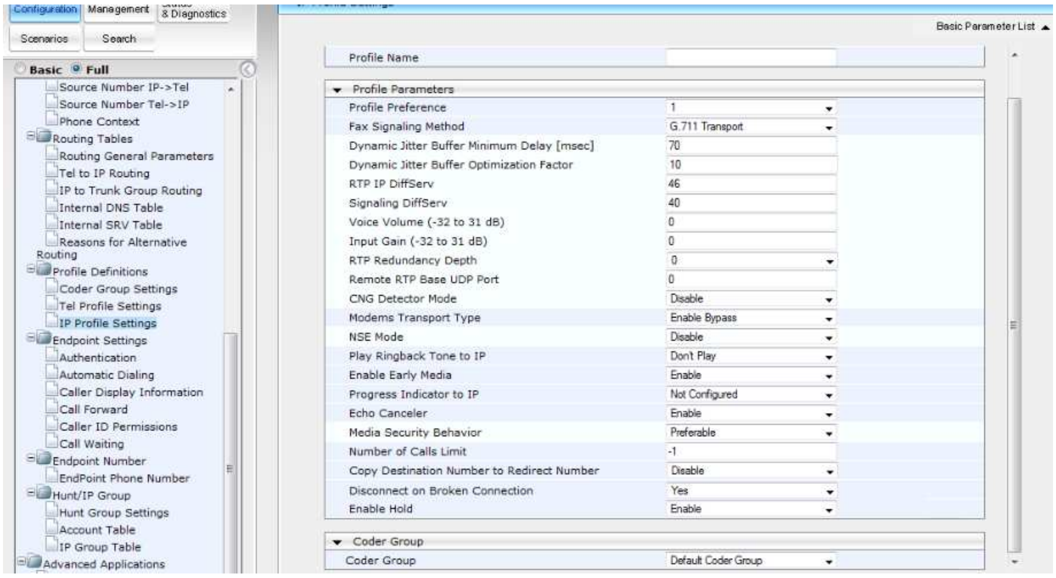

Figure 10: IP Profile configuration

4.2.2 Media architecture configuration

Select “Configuration” tab. Expand “Protocol Configuration” and “Protocol Definition” and click on “Proxy & Registration” item. “Use Default Proxy” must be set to “Yes”. OXE call server IP address must be declared as proxy in “proxy name” field.

In case of redundant architecture with two hot/standby call servers, we have to declare both OXE in proxy table. In case of failure of nominal call server, backup call server replaces the nominal one. As soon as the nominal is operational again, the backup switches back to standby mode; and the nominal becomes active. In this case, redundancy mode must be configured as “Homing”. If OXE architecture is not redundant, the redundancy mode may be set to “Parking”.

Fallback to routing table must be disabled. Therefore, to prevent any precedence of internal routing table over proxies table, “prefer routing table” field should be set to “No”.

“Always use proxy” field must be enabled so that all SIP messages and responses are sent to OXE.

MediaPack gateway needs to be registered on OXE; therefore, it is necessary to enable registration on OXE call server. In our case, maximum registration duration is set to 3600 seconds.

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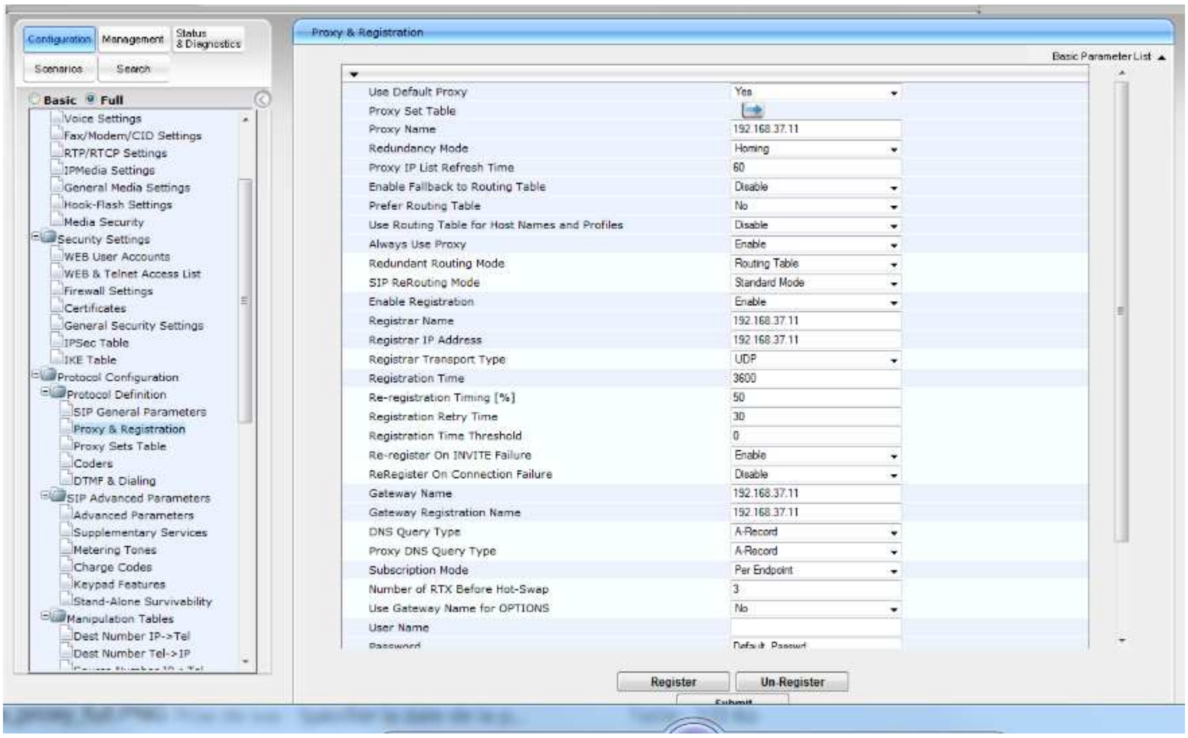

Figure 11: Proxy & Registration configuration

Select “Configuration” tab. Expand “Protocol Configuration” and “Protocol Definition” and click on “Proxy Sets Table” item in case of redundant call servers architecture. Note that in the example below, only one OXE call server is declared. OXE call servers (the nominal one in pole position followed by the backup) must be declared in the table. Calls servers are identified by their IP address and UDP transport protocol (not connected).

As OXE call servers cannot support load sharing, load balancing method must be disabled (simple or spatial redundancies architecture deployed is active/standby).

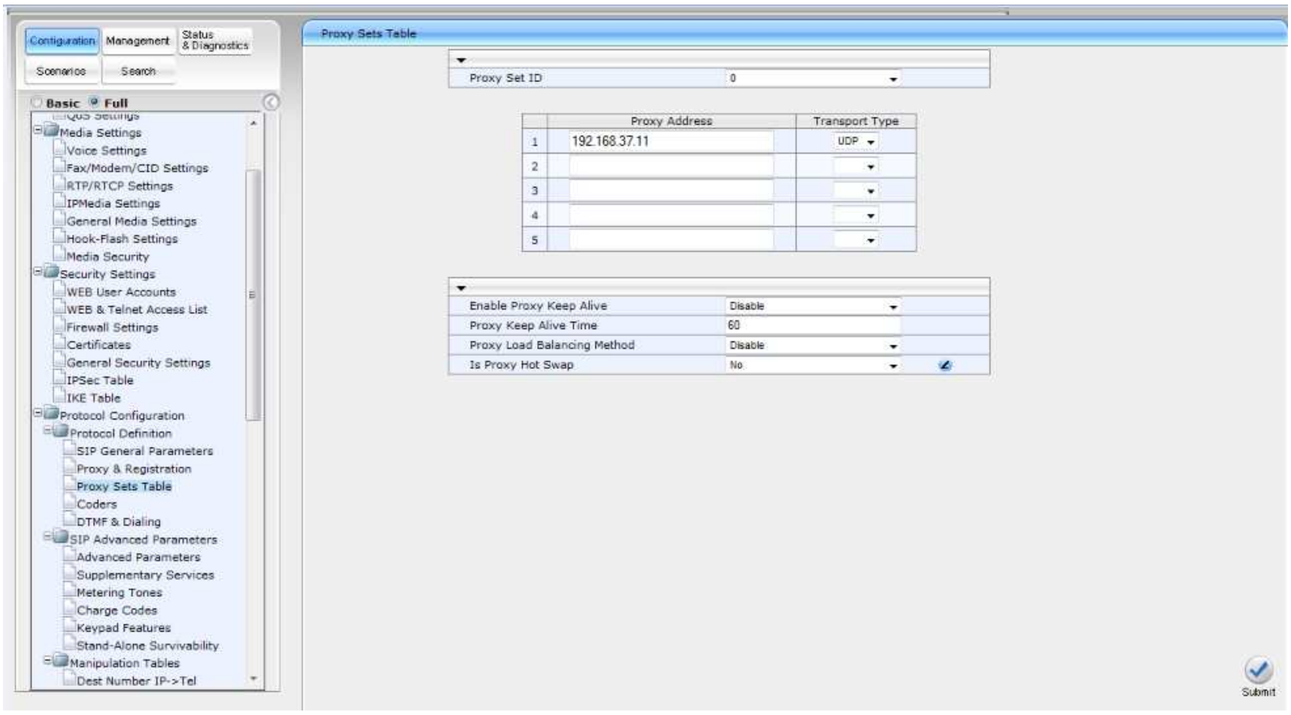

Figure 12: Proxy table configuration

4.2.3 Fax services configuration

Select "Configuration" tab. Expand "Media Settings" and click on "Fax/Modem/CID Settings" item. Fax transport mode can be set to "Disable". V.xx modem transport type must be configured as "Enable Bypass". Caller ID type should fit with ETSI standard. The codec used for fax transmissions must be G.711 A law at 64 kbps. Fax CNG Mode & CNG Detector Mode are « Disable".

USING Audiocodes MP11x FAX G711 TRANSPARENT MODE WITH OmniPCX Enterprise WITH SIP Carrier SIP Trunks : OBS BIVSIP S2

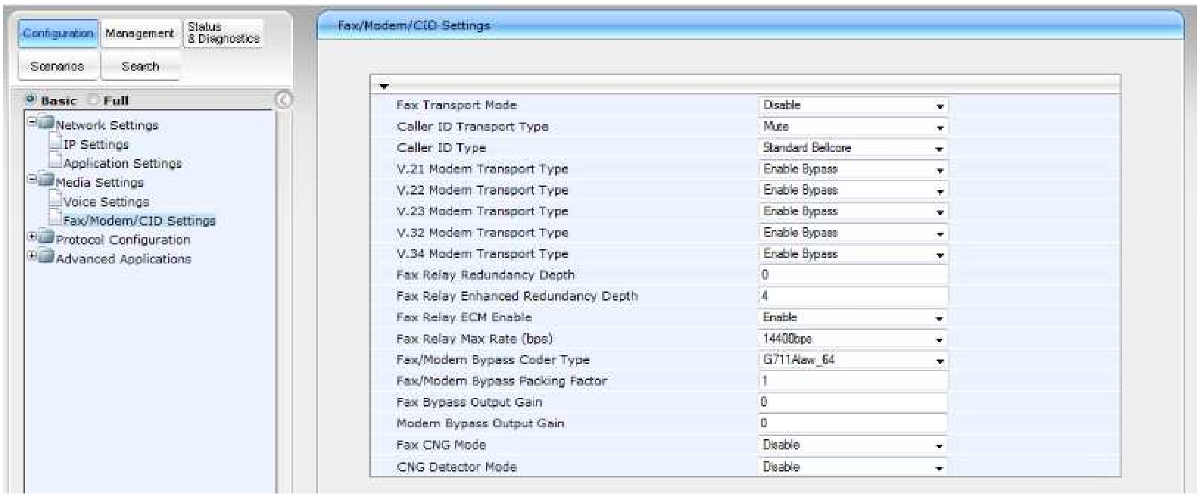

Figure 13: Fax mode configuration

Select “Configuration” tab. Expand “Media Settings” and click on “RTP/RTCP Settings” item. RFC 2833 payload types must be set to 101 in receiving as well transmitting directions. Fax bypass payload type must be set to “8”, as voice codec used is G.711A. Modem bypass payload type must also be set to “8” for same reasons.

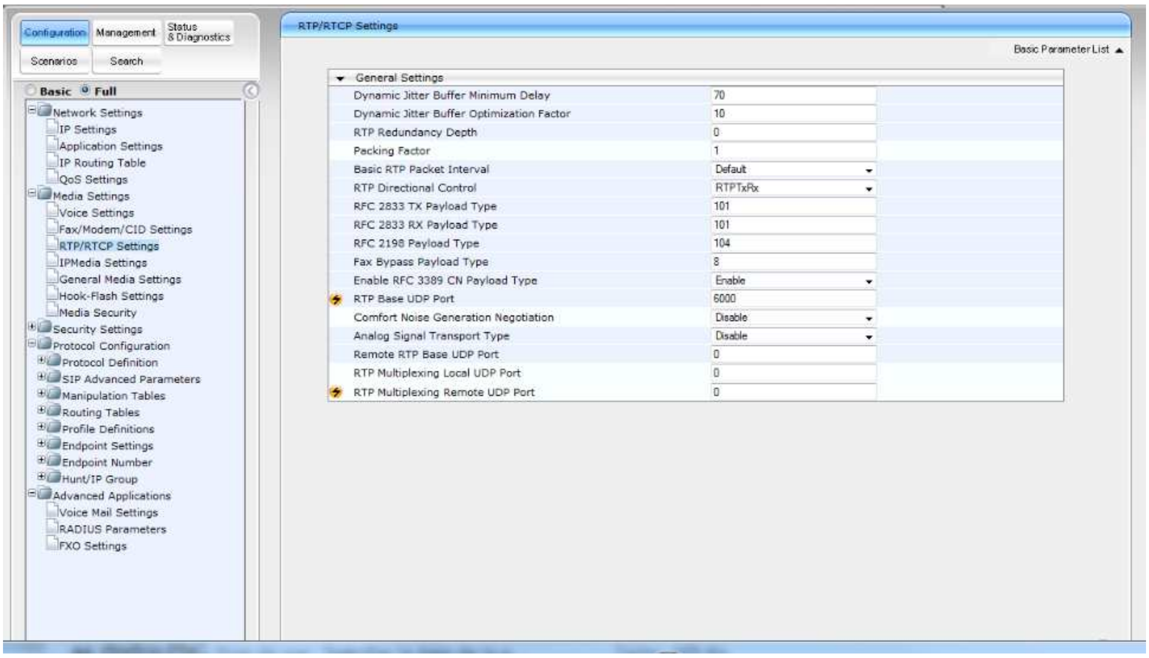

Figure 14: Fax mode configuration

As modem bypass payload type and fax/modem inband network detection are not displayed on custom graphical interface, they must be configured through the specific adminpage (http://@IP/AdminPage/). Click on “ini parameters” on left frame and declare following parameters:

- MODEMBYPASSPAYLOADTYPE = 8
- ENABLEFAXMODEMINBANDNETWORKDETECTION =1

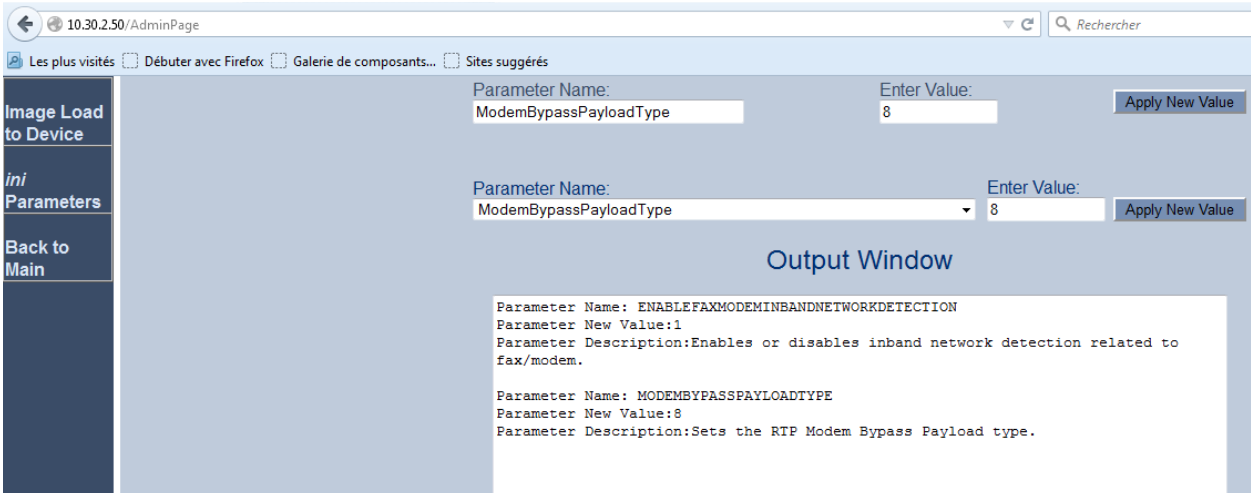

Figure 15: Specific *ini* parameters configuration

Select "Configuration" tab. Expand "Protocol Configuration" and "Protocol Definition" objects and click on "Coders" item. In order to perform fax transmissions as G.711 mode, it is needed to use G.711 A law codec (at 64 kbps with 20 ms payload) as described on following picture. Silence suppression must be disabled.

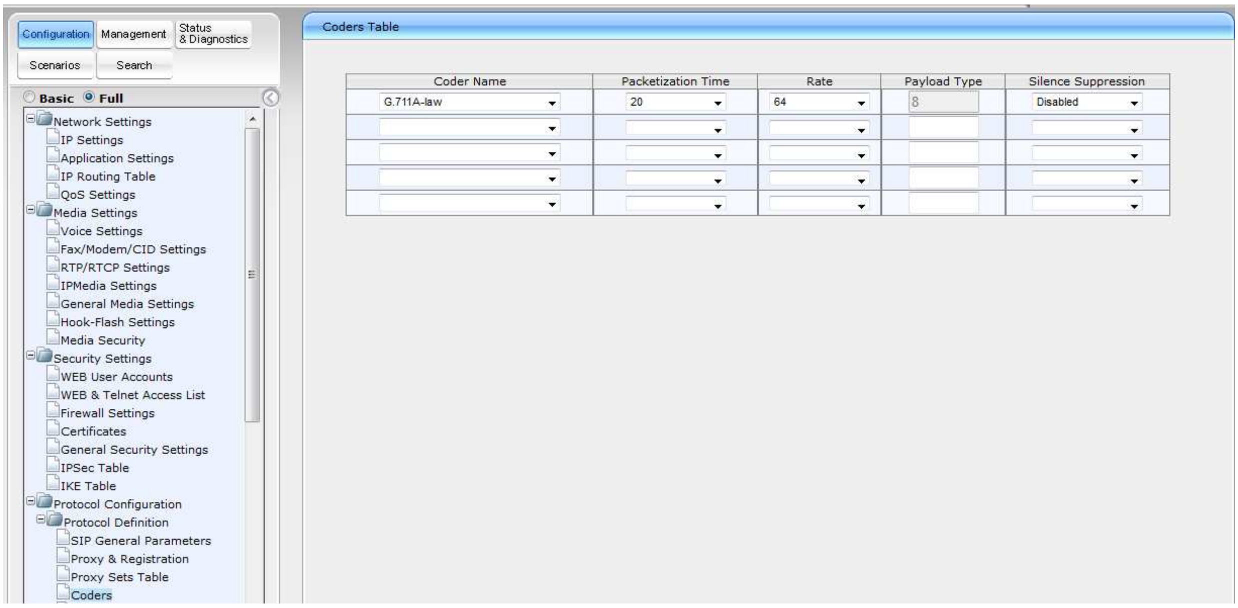

Figure 16: G711 framing configuration

Select "Configuration" tab. Expand "Profile Definition" and click on "Tel Profile Settings" item. Set up Profile Preference N°1 as below.

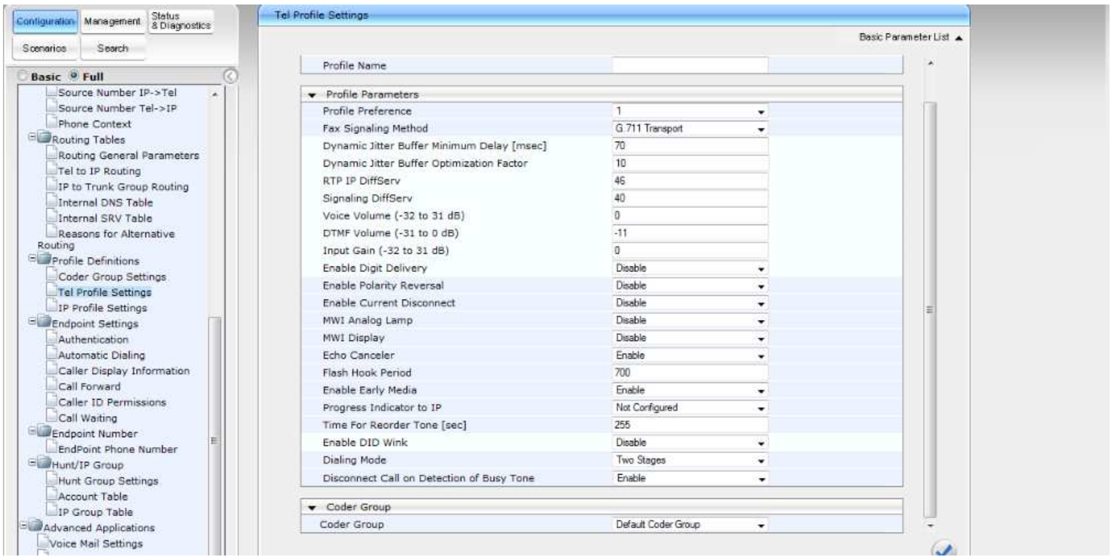

Figure 17: Tel profile for FAX configuration

Select "Configuration" tab. Expand "End point Number" and click on "End point Phone Number" item. As below, Associate MP 11x Port on which Analog Fax is connected to the Profile Preference N° defined above.

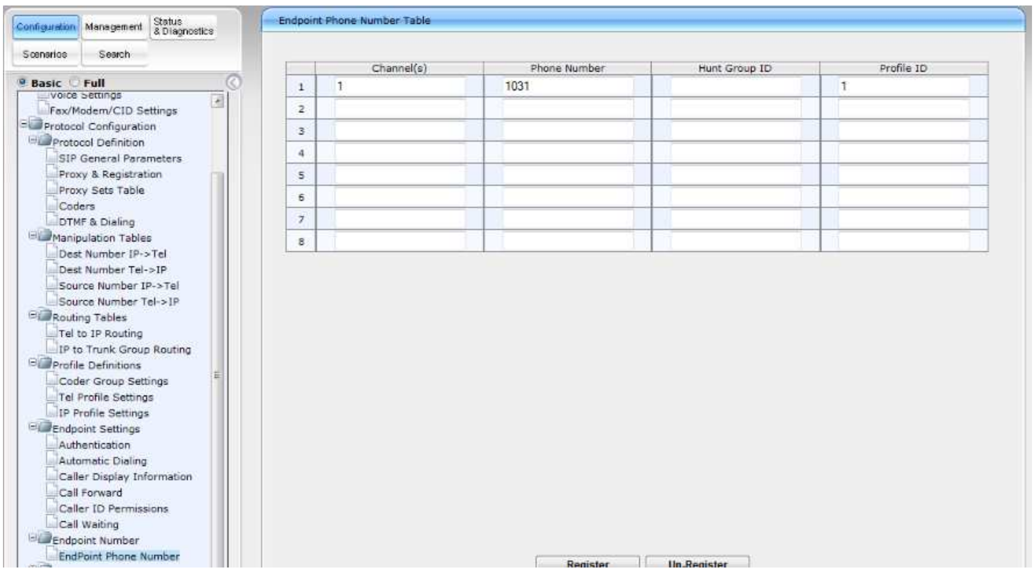

Figure 18: FAX port configuration

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