

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

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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.



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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)



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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.



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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.



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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.



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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.



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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:



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Figure 3: Incoming call

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





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:



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- 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.





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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



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



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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

	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.



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In our example, the following managements could be proposed:



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



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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 interworking 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.



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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 (<u>http://AudioCodes_IP_address</u>). 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.

Configuration Management Status		
Scenarios Search		
Basic 🍳 Full	O	
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Single IP Settings		
Single IP Settings IP Address	[10.1.10.1	
Settings Single IP Settings IP Address Subnet Mask	10.1.10.1	

Figure 7: IP Configuration



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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.

onfiguration Management Status	SIP General Parameters		
cenarios Search			Basic Parameter Li
Basic 🧟 Full	PRACK Mode	Disable 👻	
IP Routing Table	Channel Select Mode	By Dest Phone Number 🔶	
QoS Settings	Enable Early Media	Enable	
Voice Settings	183 Message Behavior	Progress 👻	
Fax/Modem/CID Settings	Session-Expires Time	0	
RTP/RTCP Settings	Minimum Session-Expires	180	
IPMedia Settings	Session Expires Method	UPDATE -	2
General Media Settings	Asserted Identity Mode	Disabled +	
Hook-Flash Settings	Fax Signaling Method	G.711 Transport 👻	
Media Security	Detect Fax on Answer Tone	Initiate T.38 on CED +	
Security Settings	SIP Transport Type	UDP +	
Protocol Configuration	SIP UDP Local Port	5060	
Protocol Definition	SIP TCP Local Port	5060	
Provi & Registration	SIP TLS Local Port	5061	
Proxy Sets Table	Enable SIPS	Disable -	
Coders	Enable TCP Connection Reuse	Enable -	
DTMF & Dialing	TCP Timeout	0	
SIP Advanced Parameters	SIP Destination Port	5060	
Advanced Parameters	Use user-phone in SIP URL	Yes 🗸	
Supplementary Services	Use user-phone in From Header	No 🗸	
Metering Tones	Use Tel URI for Asserted Identity	Disable 🗸	
Charge Codes	Tel to IP No Answer Timeout	180	
Stand-Alona Sunivability	Enable Remote Party ID	Disable •	
Manipulation Tables	Add Number Plan and Type to RPI Header	Yes 🗸	
Dest Number IP->Tel	Enable History-Info Header	Disable -	
Dest Number Tel->IP	Use Source Number as Display Name	No -	
Source Number IP->Tel	the Sceles Heese as Gallers Mashas	N1.	
Source Number Tel->IP			6
Tautian Tablas			50

Figure 8: SIP configuration (1/2)



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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".

Source Source Basic Full Basic Full Construing Table Construing Table Code Settings Voice Settings Voice Settings Voice Settings Voice Settings Voice Settings Provide Method Settings Use Display Name as Source Number Pas Display Name Control Restriction Play Ringback Tone to IP Play Ringback Tone to Tel Use Tgp Information Protocol Definition Subject Provide Settings Subject Protocol Definition Subject Provy Sets Table Coders Coders Stp Advanced Parameters Subject Settings Tomes Enable P-Charging Vector Enable P-Associated-URI Header Enable P-Associated-URI Header Subject Settings Subject Outree Parameters Enable P-Associated-URI Header Bable Order Subject Coders Enable P-Associated-URI Header Bable Order Subject Stp Advanced Parameters Enable P-Associated-URI Header Bable Order <	Lisable Vea Disable No Disable Dont Play Play According to Early IN Disable Dotte Play Play According to Early IN Disable Disable Disable AudiocodesGW	v v v v v v v v v v v v v v v v v v v	
Basic 9 Foll Basic 9 Foll IP Routing Table IP Routing Table IP Routing Table Use Source Number as Source Number	ader Yes Disable No No Disable Dont Play Pay According to Early M Disable Disable Disable AudiocodesGW	* * * * *eds * *	
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Open Settings Use Source Number as Display Name Wedia Settings Use Display Name as Source Number as Display Name Words Settings Use Display Name as Source Number as Display Name Wedia Settings Play Ringback Tone to IP Provide Settings Play Ringback Tone to IP Mode Settings Use Typi Information Booker Rash Settings Use Typi Information Security Settings SDP Session Owner Subject Subject Protocol Configuration Subject Proxy Sets Table Enable Semi-Attended Tranefer Proxy Sets Table Enable Occores DTM & Dialing Store Rawiner Advanced Parameters Enable P-Associated-URI Header Multrip Tones Forable P-Associated-URI Header Multring Tones Forable P-Associated-URI Header Protoce Configuration Subject Strain Advanced Parameters Enable P-Charging Vector Enable P-Associated-URI Header Enable P-Associated-URI Header Multring Tones Forable P-Associated-URI Header	No No Doable Dont Play Play According to Early N Deable Deable AudiocodesGW	v v Media v v	
Media Settings Use Display Name as Source Number Enable Contact Restriction Voice Settings Enable Contact Restriction Provide Settings Play Kingback Tone to TP TIPMedia Settings Use Tgri information Enable Contact Restriction Play Kingback Tone to TP Index Flags Use Tgri information Security Settings Source August Security Settings SDP Session Owner Subject Subject Protocol Configuration Subject Stp General Parameters Enable Gen-Attended Transfer Proxy Set Table Enable Concental URI Optimum Station Retry-After Time Stp Generating Y Services Retry-After Time Subject Number Preference Proxy Stat-Tary Services Sourcestad-URI Header Subject Number Preference Advanced Parameters Frable P-Associated-URI Header Matering Tones Forking Handling Mode	No Disable Dont Play Play According to Early M Disable Disable AudiocodesGW	V V Meda V V	
Voice Settings Enable Contact Restriction Fax/Modern/LDS Settings Play Ringback Tone to IP UMMedia Settings Play Ringback Tone to Tel UMMedia Settings Use Tgn information Hock-Rash Settings Use Tgn information Protocol Configuration Subject Protocol Configuration Enable Protocol Configuration Protocol Configuration Immedia Protocol Configuration Protocol Conter Enable Protocol Configuration <td>Deable Dont Play Play According to Early It Deable Deable Audiocodes/GW</td> <td>v v Meda v v</td> <td></td>	Deable Dont Play Play According to Early It Deable Deable Audiocodes/GW	v v Meda v v	
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Play Ringback Tone to Tel Diffection Settings Look-Rab Settings Look-Rab Settings Look-Rab Settings Security Settings Protocal Configuration Stp General Needia Security User-Agent Information Spottage Protocal Configuration Stp General Parameters Proxy & Registration Proxy & Registration Drox & Registration Stp General Parameters Coders Drox & Registration Stp State Table Coders Stp Advanced Parameters Advanced Parameters Retry-After Time Supplementary Services Supplementary Services Supplementary Services Proking Code	Play According to Early 1 Disable Disable AudiocodesGW	Meda v	
Use Tgrp information Enable GRUU View Tgrp information SUbject Subject Protocol Configuration Proxy & Registration Proxy & Re	Disable Disable AudiocodesGW	• •	
Hook-Flash Settings Enable GRUU Media Security User-Agent Information Security Settings SDP Security Settings Protocol Configuration Subject Protocol Configuration Subject Protocol Configuration Subject Proxy & Registration Multiple Packetization Time Format Enable Coders DTHF & Dialing Enable C-Charging Vector Enable Coders Enable C-Charging Vector Stip Advanced Parameters Retry-After Time Advanced Parameters Enable C-Associated-URI Header Supplementary Services Source Number Preference Metering Tories Forking Handling Mode	Disable AudiocodesGW		
Indekia Security User-Agent Information Security Settings SDP Session Owner Protocal Configuration SUbject Strotocal Definition Multiple Packtration Time Format Strotocal Definition Multiple Packtration Time Format Proxy & Registration Enable Semi-Attended Transfer Proxy State Table Enable Security Coders Enable PriceRel INT Strotocal Definition Multiple Packtration Time Format Proxy State Table Backering Coders Enable PriceRel INT Strotocal Definition Multiple Packtration Proxy State Table Backering Vector Enable VoiceMail URI Enable PriceRel INT StopPereneating Services Source Number Preference Matering Tomes Forking Handling Mode	AudiocodeeGW		
Security Settings Sup Session Owner Supplementary Settings Supplementary Suppleme	AudiocodesGW		
Protocol Configuration Proxy & Registration DtMr & Dialing Proxy Barameters Proxy example and Pr			
Protocol Definition SIP General Parameters Proxy & Registration Proxy Prox Proxy Proxy Proxy Proxy Proxy Proxy P			
StP General Parameters Proxy & Respiration Proxy Sets Table Coders Coders DTMF & Dialing Advanced Parameters Advanced Parameters Advanced Parameters Advanced Parameters Advanced Parameters Supplementary Services Source Number Preference Metering Tones Code Code	None	÷	100
Proxy Star Table Proxy Star Table Coders Coders DTHE & Dialing Proxy Star Table Coders Coders DTHE & Dialing Retry-After Time Advanced Parameters Retry-After Time Supplementary Services Source Number Preference Metering Tones Forking Handling Mode	Disable	¥.	
Coders	Forward	*	
DTMP & Dialing Enable VoiceMail URI Advanced Parameters Retry-After Time Advanced Parameters Enable P-Associated-URI Header Supplementary Services Source Number Preference Metering Tones Forking Handling Mode	Disable		
SID Advanced Parameters Advanced Parameters Advanced Parameters Advanced Parameters Supplementary Services Source Number Preference Metering Tones Forking Handling Mode	Disable	÷.	
Advanced Parameters Enable P-Associated-URI Header Supplementary Services Source Number Preference Metering Tones Forking Handling Mode	0		
Supplementary Services Source Number Preference Netering Tones Forking Handling Mode	Disable	¥	
Metering Tones Forking Handling Mode			E
Charge Codes	Sequential handling		
Enable Reason Header	Enable	•	
Keypad Features			
Stand-Alone Survivability Retransmission Parameters			
Dest Number IP->Tel	500		
Dest Number Tel-> IP	4000		
Source Number IP->Tel	7		

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



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enarios Search				Dasic Paramete
Insie 9 Full	Profile Name			
Source Number IP->Tel	Profile Parameters			
Source Number Tel->IP	Profile Preference	1		
Phone Context	Fax Signaling Method	G.711 Transport		
Routing Tables	Dynamic Jitter Buffer Minimum Delay [msec]	70		
Routing General Parameters	Dynamic Jitter Buffer Optimization Factor	10		
Tel to IP Routing	RTP IP DiffServ	46		
Internal DNS Table	Signaling DiffServ	40		
Internal SRV Table	Voice Volume (-32 to 31 dB)	0		
Reasons for Alternative	Input Gain (-32 to 31 dB)	0		
Louting	RTP Redundancy Depth	0		
Profile Definitions	Remote RTP Base UDP Port	0		
Coder Group Settings	CNG Detector Mode	Disable		
Tel Profile Settings	Modems Transport Type	Enable Bypass	-	
Endpoint Settings	NSE Mode	Disable		
Authentication	Play Ringback Tone to IP	Don't Play		
Automatic Dialing	Enable Early Media	Enable		
Caller Display Information	Progress Indicator to IP	Not Configured	*	
Call Forward	Echo Canceler	Enable	-	
Caller ID Permissions	Media Security Behavior	Preferable	•	
Call Waiting	Number of Calls Limit	-1		
Endpoint Number	Copy Destination Number to Redirect Number	Disable		
Hust/IB Group	Disconnect on Broken Connection	Yes	+	
Hunt Group Settings	Enable Hold	Enable	-	
Account Table		570000 U		
IP Group Table	- Coder Group			

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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statement in the state				Basic Paramete
Scenarios Search	•	1000000		
Basic 9 Full	Use Default Proxy	Yes	•	
Voice Settings	Proxy Set Table			
Fax/Modem/CID Settings	Proxy Name	192.168.37.11		
RTP/RTCP Settings	Redundancy Mode	Homing		
DIPMedia Settings	Proxy IP List Refresh Time	60		
General Media Settings	Enable Fallback to Routing Table	Disable	-	
Hook-Flash Settings	Prefer Routing Table	No		
Media Security	Use Routing Table for Host Names and Profiles	Disable		
Security Settings	Always Use Proxy	Enable		
WEB User Accounts	Redundant Routing Mode	Routing Table		
WEB & Telnet Access List	SIP ReRouting Mode	Standard Mode		
Firewall Settings	Enable Registration	Enable	-	
Constal Ecourity Cottings	Registrar Name	192.168.37.11		1
Dipsec Table	Registrar IP Address	192 168 37 11		
	Registrar Transport Type	UDP		
Protocol Configuration	Registration Time	3600		
E Protocol Definition	Re-registration Timing [%]	50		
SIP General Parameters	Registration Retry Time	30		
Proxy & Registration	Registration Time Threshold	0		
Proxy Sets Table	Re-register On INVITE Failure	Enable		
	RePerigter On Connection Failure	Disabla		
DTMF & Dialing	Galaviav Name	192 168 37 11		
SIP Advanced Parameters	Gateway Name	192 169 37 11		
Advanced Parameters	DNC Queer Tree	A Decord	10	
Matorina Tasas	Diris Query Type	6 Decend		1,3
Charge Codes	Proxy DNS Query Type	Pro Code stat		
Keypad Features	Subscription Mode	Per Endports		
Stand-Alone Survivability	Number of KIX before hot-Swap	3		
C Manipulation Tables	Use Gateway name for OP110NS	190		
Dest Number IP->Tel	user name	D.(.). D		
Dest Number Tel->1P	Vaseworn	runa di Panouri		64 C
Parana Branchas 10 a Wat	Desi	In Pasistar	1	
	Regi	ster on Register	4:	

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).



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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".



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ionfiguration Management Status & Disgnostics	Pax/Modem/CIU-Settings		
Scenarios Search	[
Basic O Full	Fax Transport Mode	Disable	
P Network Settings	Caller ID Transport Type	Mute	-
IP Settings	Caller ID Type	Standard Belicore	-
Application Settings	V.21 Modern Transport Type	Enable Bypass	•
Media Settings	V.22 Modern Transport Type	Enable Bypass	+
Voice Settings	V.23 Modern Transport Type	Enable Bypass	*
Enternal Configuration	V.32 Modern Transport Type	Enable Bypass	•
# Advanced Applications	V.34 Modern Transport Type	Enable Bypass	•
	Fax Relay Redundancy Depth	0	
	Fax Relay Enhanced Redundancy Depth	4	
	Fax Relay ECM Enable	Enable	-
	Fax Relay Max Rate (bps)	14400bpe	* Îl
	Fax/Modern Bypass Coder Type	G711Aaw_64	•
	Fax/Modern Bypass Packing Factor	1	
	Fax Bypass Output Gain	0	
	Modern Bypass Output Gain	0	
	Fax CNG Mode	Disable	÷jil .
	CNG Detector Mode	Disable	-

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.

	✓ General Settings			
Basic [®] Full	Dynamic Jitter Buffer Minimum Delay	70	li i i i i i i i i i i i i i i i i i i	
Network Settings	Dynamic Jitter Buffer Optimization Factor	10		
IP Settings	RTP Redundancy Depth	0		
Application Settings	Packing Factor	1		
IP Routing Table	Basic RTP Packet Interval	Default	+	
Qos Settings	RTP Directional Control	RTPTxRx	•	
Voice Settings	RFC 2833 TX Payload Type	101		
Fax/Modem/CID Settings	RFC 2833 RX Payload Type	101		
RTP/RTCP Settings	RFC 2198 Payload Type	104		
IPMedia Settings	Fax Bypass Payload Type	8		
General Media Settings	Enable RFC 3389 CN Payload Type	Enable	•	
Hook-Flash Settings	😸 RTP Base UDP Port	6000		
Media Security	Comfort Noise Generation Negotiation	Disable	*	
Security Settings	Analog Signal Transport Type	Disable	•	
Protocol Configuration	Remote RTP Base UDP Port	O		
Basip Advanced Parameters	RTP Multiplexing Local UDP Port	0		
* Manipulation Tables	RTP Multiplexing Remote UDP Port	0		
SiP Advanced Parameters Manipulation Tables Profile Definitions Profile Definitions File Analysis Endpoint Settings Munt/1P Group Advanced Applications Voice Meil Settings	RTP Multiplexing Lecal UDP Port	0		f

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 (<u>http://@IP/AdminPage/</u>). Click on "ini parameters" on left frame and declare following parameters:

- MODEMBYPASSPAYLOADTYPE = 8
- ENABLEFAXMODEMINBANDNETWORKDETECTION =1



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🗲 🚭 10.30.2.50/AdminPage		⊽ C Q Rechercher	
Les plus visités Débuter avec Firefox Galerie de composant	s 🗌 Sites suggérés		
Image Load to Device	Parameter Name: ModemBypassPayloadType	Enter Value: Appl 8	y New Value
ini Parameters	Parameter Name: ModemBypassPayloadType	Enter Value:	y New Value
Back to Main	Ou	tput Window	
	Parameter Name: ENABLEFAXMODEMINBA Parameter New Value:1 Parameter Description:Enables or di fax/modem.	IDNETWORKDETECTION	to
	Parameter Name: MODEMBYPASSPAYLOAD Parameter New Value:8 Parameter Description:Sets the RTP	YPE Modem Bypass Payload type.	

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.

Basic 9 Full		Packetization Time	Rate	Payload Type	Silence Suppression
	G.711A-law	20 🗸	64 👻	8	Disabled -
Network Settings			()		
IP Settings		·	· · · · · · · · · · · · · · · · · · ·		· · · · · · · · · · · · · · · · · · ·
Application Settings	-	▼	· · · · · · · · · · · · · · · · · · ·		v
IP Routing Table	•		· · · · ·		•
QoS Settings			1		
Media Settings					• •
Hook-Flash Settings Media Security Security Settings WEB User Accounts WEB & Telnet Access List Firewall Settings					

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.



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and frame			Basic P	arameterList 🔺		
ands Search	Profile Name					
Basic 🖲 Full 🕜	Traine Hume			_		
Source Number IP->Tel + Source Number Tel->IP	Profile Parameters					
	Profile Preference	Profile Preference 1 -				
Phone Context	Fax Signaling Method	G.711 Transport	•			
Routing Tables	Dynamic Jitter Buffer Minimum Delay [msec]	70				
Routing General Parameters	Dynamic Jitter Buffer Optimization Factor	10				
The Touck Course Pourling	RTP IP DiffServ	46				
Internal DNS Table	Signaling DiffServ	40				
Internal SRV Table	Voice Volume (-32 to 31 dB)	0				
Reasons for Alternative	DTMF Volume (-31 to 0 dB)	-11				
Routing	Input Gain (-32 to 31 dB)	0				
Profile Definitions	Enable Digit Delivery	Disable	+			
Coder Group Settings	Enable Polarity Reversal	Disable				
Tel Profile Settings	Enable Current Disconnect	Disable				
IP Profile Settings	MWI Analog Lamp	Disable		=		
Authentication	MWI Display	Disable				
Automatic Dialing	Echo Canceler	Enable				
Caller Display Information	Elash Hook Period	700				
Call Forward	Enable Early Media	Enable				
Caller ID Permissions	Progress Indicator to IP	Not Configured	-			
Call Waiting	Time For Reorder Tone [sec]	255				
Endpoint Number	Enable DID Wink	Disable	-			
EndPoint Phone Number	Dialing Mode	Two Stages				
Hunt/IP Group	Disconnect Call on Detection of Buck Tone	Enable				
Hunt Group Settings	Lasconnect can on Detection of busy rone	Crisbie	•	-		
IR Crown Table	Coder Group					
Advanced Applications	Coder Group	Default Coder Group	-	1.12		

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.

Management & Diagnostics							
cenarios Search							
		Channel(s)	Phone Number	Hunt Group ID	Profile ID		
Basic C Full	1	1	1031		1		
Voice Settings	2						
Protocol Configuration	-						
Protocol Definition	3						
SIP General Parameters	4						
Proxy & Registration	5						
Proxy Sets Table	-						
Coders	100						
DTMF & Dialing	7			1			
Manipulation Tables	8						
Tel to IP Routing Tel to IP Routing IP to Trunk Group Routing Profile Definitions Coder Group Settings Tel Profile Settings Profile Settings Authentication Automatic Dialing Caller Dialing Caller Dialing							
Call Forward Caller ID Permissions							
Call Forward Caller ID Permissions Call Waiting Endpoint Number							

Figure 18: FAX port configuration



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